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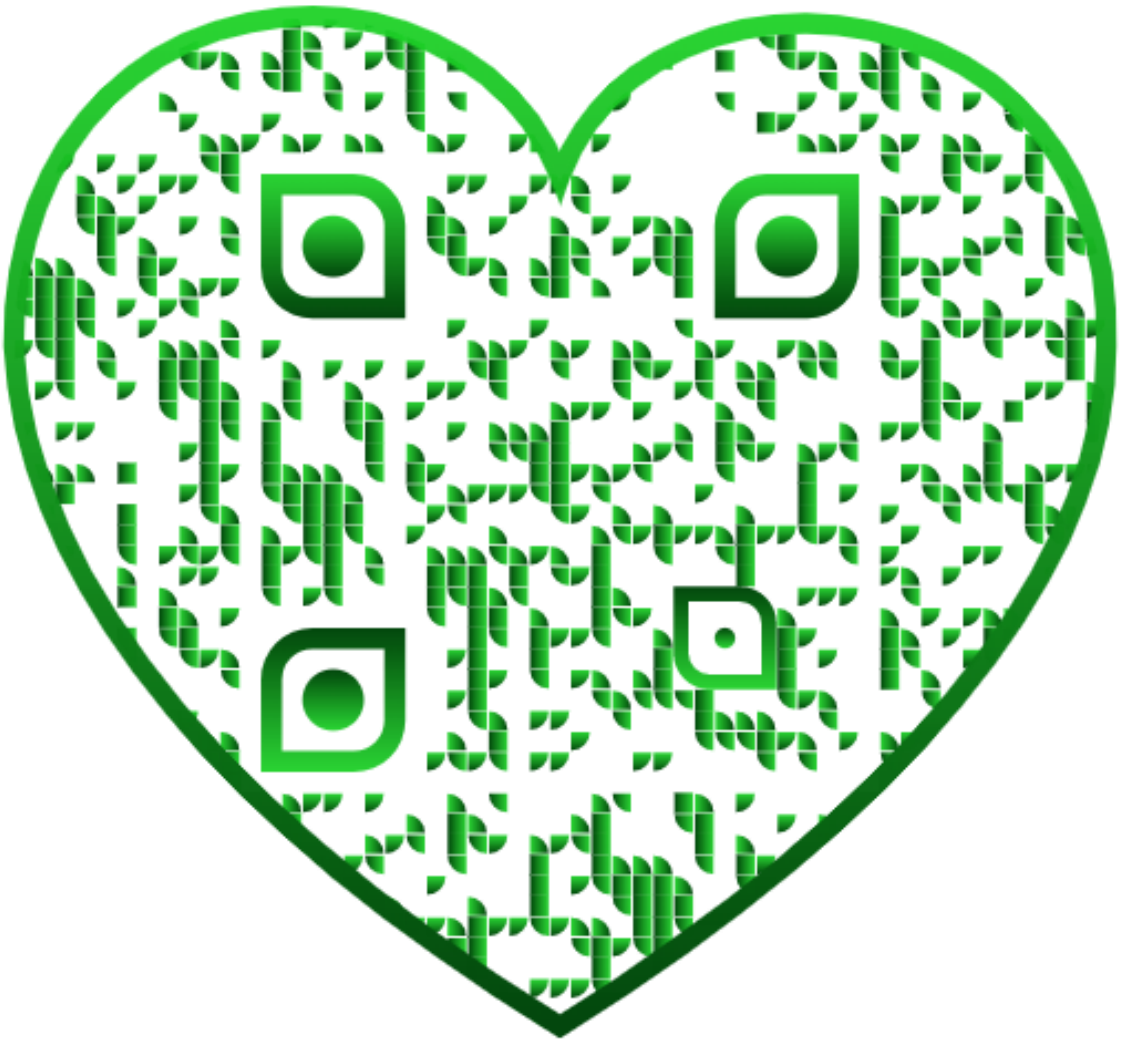

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SEIZE THE MOMENT

Our perspective on studio gear is often a romantic one. We operate in the conviction that, somewhere, there exists a microphone that will perfectly complement our singing voice, or a synth that fits precisely with our musical vision. The quest to find these touchstones is a constant backdrop to our musical endeavours, and lends an air of permanent compromise to the setups we use along the way.

The same affliction plagues guitarists, with their ever-changing pedalboards. It sees modular enthusiasts coming a mile off. I'd suggest it's almost a universal feature of music-making today — except that the world of live sound seems largely immune. We all care about sounding good and get frustrated when we sound bad, or can't hear ourselves properly. Many of us no doubt have a favourite stage vocal mic, or a general preference for in-ears versus wedges. But there's rarely the same perfectionism or willingness to dive into technical rabbit holes that you find in the studio world.

There are obvious reasons for this. In live sound, we're often dependent on unknown sound engineers, venue equipment, room acoustics and so on. There's not much point in chasing the last five percent when so much of the first 95 percent is out of our hands. And in a live context we're more inclined to stick to the tried and the tested, since the penalties for getting it wrong can be disastrous.

But things are changing. I recently recorded a live session for a band who had ditched physical guitar and bass amps, bought their own vocal and drum mics, and invested in a rackmount digital mixer and IEMs for everyone. The idea is that they can turn up at any venue and give the sound engineer exactly the sounds they choose — even just a stereo mix, if needs be. I confess I went into the session thinking 'Isn't this all going to sound a bit sterile? Where will the energy come from?'

I came out of the session thinking 'Why isn't everyone doing this?' From a recording perspective, what the band had done was take control over probably the single most important factor in delivering a good performance: their cue mix. You can't completely guard against the damage that an inept house engineer or poor PA installation can do to your live sound. But if you go a little way down the rabbit hole, you can take control over a huge swathe of that 95 percent. Stageboxes and IEMs might not be as glamorous as vintage synths or mics, but they can take you into every gig confident that you'll be able to deliver your best possible performance. And that, surely, is where the real romance lies.

Sam Inglis
Editor In Chief

"You can't completely guard against the damage that an inept house engineer or poor PA installation can do to your live sound."

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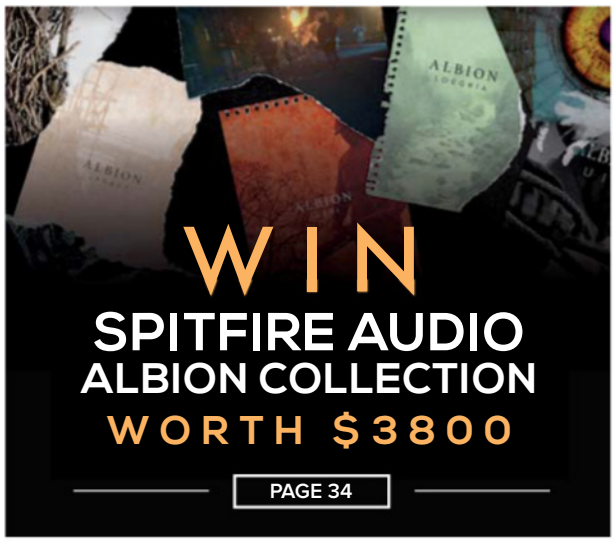

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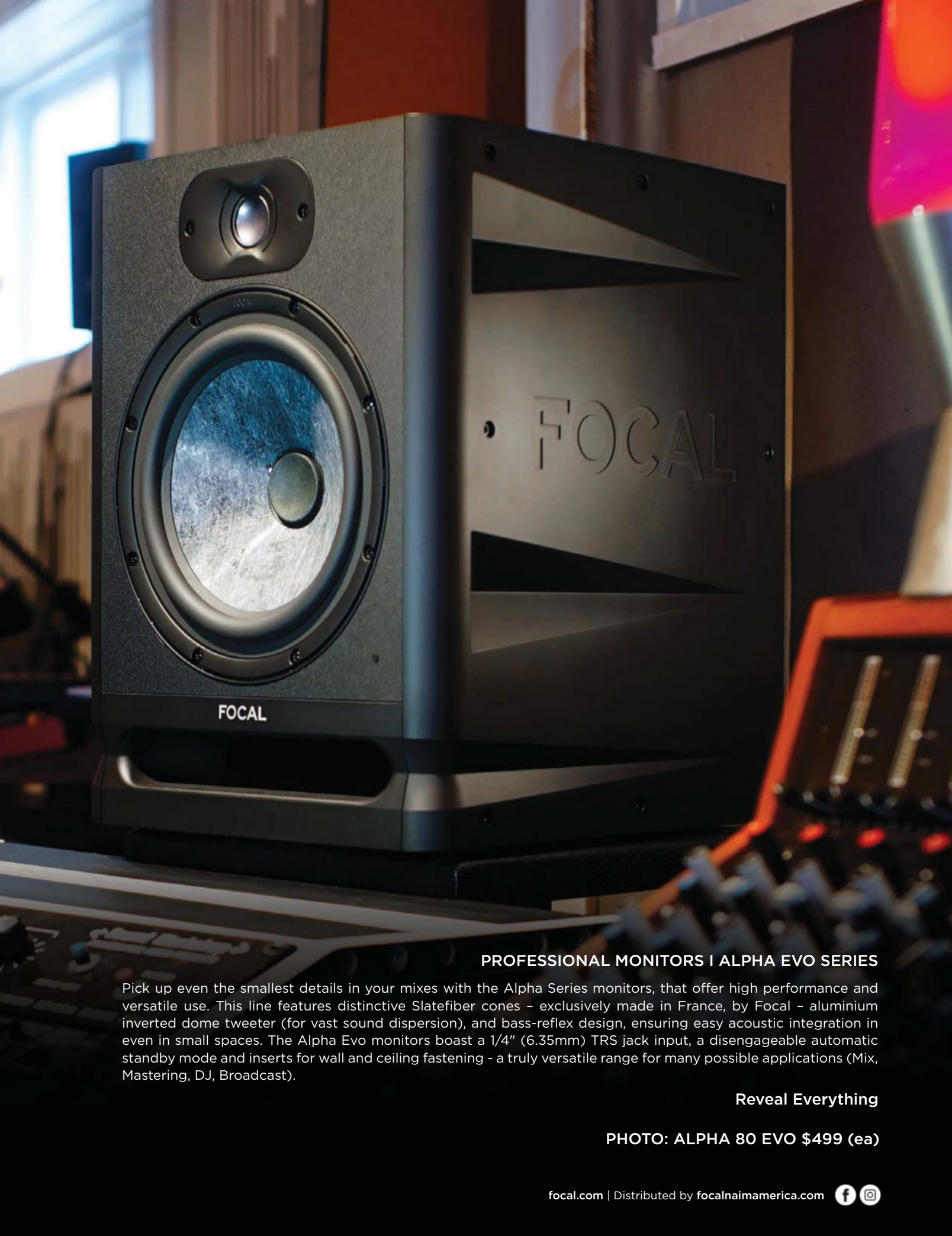

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Waves Voltage Amps

Amp-modelling Plug-in

With seven amp and six cabinet emulations, there's a wide range of guitar and bass tones on offer here.

PAUL WHITE

The new Voltage Amps has options for bass and electric guitar, and is available in the usual plug-in formats for Mac (Intel or Silicon, running macOS Catalina 10.15 or later) and Windows. It seems pretty light on CPU — it barely tickles the CPU load meter on my M2 Mac mini. Separate plug-ins are provided for the guitar and bass plug-ins, with mono and stereo versions of each. The emphasis is on making it easy to dial up the desired sounds quickly, and to that end it has a special feature called Focus, the aim of which is to help match your guitar and its pickup types to the amps on offer.

Nuts & Volts

The plug-in is almost as quick and easy to tweak as an actual amp, but there are also plenty of presets — over 400, in fact, many created by leading musicians and producers. As you might expect, the amps are informed by classic UK and US types, and the sounds range from super clean to aggressively distorted, but while they cover the classic tonal colours you might expect from Fender, Marshall, Mesa Boogie and high-gain models such as Soldano, these models don't purport to be precise emulations of specific amplifiers.

Silverado excels at clean tones, while Arena is good for serving up stadium rock. Royal-X's jangly tone hints in the general direction of Vox or Matchless, while Blue Flame goes for a boutique amp sound. For searing metal tones, the aptly named Aggro will oblige. For bass players, the two amp models go from smooth and deep to vintage tones. Vintage Velvet sounds similar to what you might expect from an old Ampeg, while Dark Mass delivers a more solid, assertive, modern bass sound.

Although each amp has a different look, their control layouts are all identical. A global Input control adjusts the level being fed to the amp models, and this has a small meter with a clip indicator. The selected amp's controls comprise Focus, Volume, Gain, a three-band EQ and a noise gate with variable threshold and time, and the amp can be set to Clean, Overdrive, or Lead mode. The Focus control helps compensate for your guitar's pickup types, which can vary from bright single coils to thick-sounding humbuckers; essentially, you just turn the control until you like what you hear.

The amp can be paired with one of six IR-based cab setups, and each selection comprises two IRs, with a Dark/Bright knob setting the balance between them. For when you're using a separate cab emulator or feeding a physical amp and speaker, the cab IR stage can be bypassed. When the Lock switch to the left of the EQ is engaged, each amp's individual settings are maintained when you change from one amp to another or choose another channel type. A further dial adds stereo room ambience, which creates a natural sense of space (if heard without added reverb, especially over headphones, modelled amps tend to sound very dry, and nobody records in an anechoic chamber!).

Verdict

Each amp has its own sonic character, and even the high-gain amps have a clean setting, which affords a wide range of clean tones. To my ears, the drive sounds have what I'd call a 'US character', which is less smooth than what you'd expect from UK classic rock. But if you find the driven sounds too gritty for your taste, adding some steep top-cut EQ from another plug-in is a quick fix. The Arena amp comes closest to what I'd expect a UK rock amp to sound like, and is capable of some great lead sounds. However, if gentle blues-style dirt is more your thing, you may find that there's a gap between the sound you get using high gain in the Clean setting and low gain in the Drive position — if the low-drive settings are still too dirty, you can reduce the input level so that less signal is fed into the amp. There are no spring reverbs or tremolos, but that's no biggie in my view, as the room ambience is decent, and we all have tremolo and reverb plug-ins in our DAWs.

The Focus control comes in very handy if you need to warm up the sound of bright pickups, but it's also good for adding definition to thick-sounding humbuckers. You can also use your own overdrive pedals with the amps set to clean, for an even wider range of tones. It is also worth experimenting with any other cabinet models you have; you'd normally bypass the internal cabs when doing this, but you can sometimes get some useful results with both active. Finally, as always with amp sims, remember that they're not just for guitar: Voltage Amps sounded great for thickening or dirtying drums and adding attitude and warmth to synths. All in all, then, it's a versatile plug-in that sounds good and is simple to operate — well worth trying. ■■■

summary

A versatile and really easy-to-use amp-modelling plug-in with a wide tonal palette and some interesting features.

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The m908 Monitor Controller gets you working in any format quickly and easily, expertly managing all speaker systems from stereo to Dolby™ Atmos 9.1.4. New firmware 2.0 includes a web-based control platform which lets you operate and configure the entire system from any web browser (desktop or mobile). Additionally, the m908's room correction EQ capability has been tweaked to provide 12 bands on all 24 channels at all sample rates. These updates, combined with its unrivaled audio performance and mechanical elegance, cement the m908 as the finest, all-in-one monitoring tool for modern music production.

"I find a monitor controller to be an indispensable part of studio life these days, particularly in a console-less room without a master section for speaker/input switching, and now with multi-channel mixing like Atmos becoming more prevalent, a properly built monitor controller for that scenario is vital. The Grace Design m908 absolutely fits that description; incredible sound quality, options for almost any input and output need, the flexibility to program multiple speaker layouts and have them available at the press of a button are just a few of the top features in a truly full-featured unit."

Don Gunn, TapeOp Magazine

GRACE DESIGN

www.gracedesign.com

Bereich03 Audio BAX-EQ

500-series EQ

Each band of this mastering-grade module can process the Mid, Sides or stereo signal.

BOB THOMAS

The two-band Baxandall tone control that crops up in millions of hi-fi units must rank as the greatest gift ever given to the audio industry! Until relatively recently, Baxandall's signature achievement appears to have been virtually ignored by the professional recording industry but, fortunately, over the last decade or so, a few boutique pro-audio manufacturers have sought to rectify that situation. Among these is Michael Heilrach's Munich-based Bereich03 Audio. Focused entirely on 500-series-format mastering equipment, the company have earned a considerable reputation for the functionality, quality and cost-effectiveness of their products — as evidenced by their reviews in this very magazine. Bereich03's most recent release, the BAX-EQ, is a 500-series, Baxandall-based mastering EQ, each of whose three bands can be switched individually between Mid-only, Sides-only and stereo operation.

Overview

Featuring Bereich03's signature cosmetics and engraved model designation, the BAX-EQ is a superbly built double-width module. The audio performance is impressive, including a frequency response that's flat from 5Hz until it drops by -1dB at 199kHz, a noise level of -113dBA, and a maximum input/output level of +24dBu. All the rotary controls are selector switches, and a solitary LED illuminates when the BAX-EQ has been activated, and taken out of hard bypass.

Controls for each band are identical. A dial on the left sets the band's corner frequency, and between it the gain control on the right sit two toggle switches, the upper selecting between Mid, Sides and full stereo operation, and the lower setting the gain function to boost or cut mode. Two gain configurations are available: ±5db in 1dB steps, and ±3dB in 0.5dB steps. As you'd expect, the high and low bands have a shelving response that appears to be classic Baxandall, whilst the mid band has a bell-shaped response with a wide Q and relatively gentle frequency response curve that gives you the ability to finesse the midrange. The corner frequency choices for each band are decidedly individual and, I would imagine, drawn from Michael Heilrach's long experience as a mastering engineer. The high-shelf frequencies are 1.2, 2.5, 4.7, 8, 11 and 18 kHz; the midrange's are 190, 340, 540, 770 and 1100 Hz; and the low shelf offers 35, 47, 62, 82, 120 and 210 Hz.

In Use

I inserted the BAX-EQ into my master bus after my Tegeler Audio Crème mastering compressor, which features a passive, boost-only, two-band Pultec-style EQ with shelving treble and bass. The first thing I noticed is that operating the BAX-EQ's switches produces low-level transients, some of which can be audible — not a big deal for me, as I don't change master bus compression and EQ within a track, but your workflow may differ. Other than that, when set flat and whether in or out of circuit, the BAX-EQ is completely transparent: it adds and subtracts nothing.

As a stereo two-band equaliser, the BAX-EQ has a much wider high-frequency coverage than the Tegeler and, although I preferred the BAX-EQ overall, there were tracks where the Pultec's more granular approach worked better. Add in the BAX-EQ's additional mid band and Mid-Sides capabilities, and the contest is over... though combining the two EQs can create serendipitous results! The BAX-EQ is intuitive to use, and after working on a few masters, stem projects and complete mixes it didn't take me long to develop what now serves as my starting point: a 3dB, 82Hz low-frequency Mid boost; a 1dB, 1.1kHz midrange Sides cut; and a 2dB, 8.0kHz high-frequency stereo boost.

The ability to switch each of its bands into either Mid, Sides or stereo mode offers significant sonic benefits that can't be found elsewhere and, given its audio performance, feature set and price, the Bereich03 BAX-EQ offers great value for money. Purchasing one is, to me, a no-brainer: this example is not going anywhere! ■■■

summary

A three-band, mastering-grade equaliser for the 500 series, the BAX-EQ is based on the Baxandall design and boasts an audio performance of the highest quality. It also has the unique ability to run each band in Mid, Sides or stereo modes.

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UJAM Usynth GLAM

Virtual Synthesizer Instrument

UJAM's Usynth range is about getting good results quickly, and GLAM is no exception.

JOHN WALDEN

Alongside their virtual guitarists, bassists and drummers, UJAM have recently added synth-based instruments to their catalogue with their Usynth line. The latest addition to this range is GLAM and, like the other Usynth titles, GLAM's sounds are designed to suit a particular musical style. In this case, that sonic target is polished, modern, electronic pop. Like all UJAM's products, Usynth's design is intended to give the user high-quality sounds within a streamlined control set. The aim is to avoid getting bogged down in option paralysis and, instead, to get on with your creative process. Therefore, the UI is stylish, simply laid out, and not overly stuffed with endless synth parameters. Don't be fooled, though; Usynth GLAM gives you plenty of sound-design options and, given how good many of the presets sound, there appears to be a lot going on under the hood.

More Than Meets The Eye

While the detailed cosmetics change, GLAM shares the same UI with other Usynth titles. As such, the user gets access to presets along the top, the Sequencer, Synthesizer and Finisher (a multi-effects engine) control panels (each with their own additional preset systems) and, at the base, Delay and Ambience effects, a virtual keyboard and the excitingly named Surprise section. This last offers a whole range of randomisation options; it's a very clever design, allowing you to generate subtle variations or dramatic changes in the current sound.

The Usynth engine includes a number of different synthesis types, and each preset uses a specific combination of these. However, as the UI makes obvious, the Synthesizer section provides just a few controls. The Dark/Bright knob serves as a filter, while Fast/Slow changes the sounds amplitude envelope. As with almost all the controls, these macro-style controls often tweak multiple parameters behind the scenes. The labels on the smaller Synthesizer controls change based upon the starting preset and are design to provide the most obvious sound-design options for that preset. The same macro control approach applies in the Finisher section, so you can pick a preset starting point for the style of processing you want and then tweak from there. Overall, the UI is a breeze to use but still offers enough control to let you dial sounds in.

"For producers who just want a palette of top-notch, easily tweaked synth sounds but without having to finish their PhD in synthesis, GLAM may well appeal."

While the control set is streamlined, the sounds are not — GLAM is certainly capable of packing a sonic punch. The presets are divided into various categories including basses, pads, leads, stabs, melodies, arpeggios and ear candy. Many of these make use of the Sequencer section and this offers various modes of operation with presets for arpeggios, chords, sequences and motifs. There are options to adjust these presets, and these include keyswitchable pattern styles mapped from C1 to B1. Perhaps the only limitation here is that you can't edit the patterns or (as far as I could see) export the MIDI produced by a pattern for subsequent editing within your DAW.

Conclusion

That comment aside, for producers who just want a palette of top-notch, easily tweaked synth sounds but without having to finish their PhD in synthesis, GLAM may well appeal. And if that might be you, then you can try GLAM or any of the other Usynth titles for yourself via UJAM's free 30-day trial.

summary

With UJAM's streamlined design ethos underpinning the Usynth front-end, GLAM provides a great collection of synth sounds for electronic pop, for producers who would rather avoid complex synth engines.

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Hugh Robjohns - Sound On Sound

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PSPaudioware PSP42 v.2

Modulated Delay Plug-in

We check out the latest version of PSP's Lexicon PCM42 emulation.

PAUL WHITE

PSP42 was one of PSPaudioware's earliest plug-ins, and back in the day acquired an enviable reputation, but a revision was arguably overdue. In this new version, PSP have improved the accuracy of their modelling, while also adding some practical feature enhancements. Lexicon's PCM42 delay, which this plug-in models, was valued both for its versatility and its musical, non-clinical sound. Its modulation section comprised an LFO that offered sine or square waveforms plus an envelope follower, which could be mixed with the LFO modulation signal to affect the delay time and pitch. For its time, it was a very impressive device — a multipurpose delay that could also be used as a phrase sampler, and which, by changing the delay time (by altering the sampling rate), could deliver tape-like pitch changes too.

Features

PSP42 v.2 models the audio path of the PCM42 using one of three operating modes. Clean bypasses the signal path emulation, while Lims brings in emulations of the two optical limiters used in Lexicon's original circuit, one at the input and the other immediately before the A-D converter stage. The first works on both the dry and wet signals, while the second affects just the wet and feedback signals. Full mode again brings the limiters into play but also replicates the coloration imparted by the original A-D and D-A converters. Legacy engages the pre-A-D soft saturator that was found in the original PSP42 plug-in.

The enhancements include an increased delay time (now up to 28.8 seconds). PSP have also added a high-cut filter, LFO tempo sync and adjustable input level sensitivity, to emulate -10dBV or +4dBu operation. Other additions include the ability to invert the envelope controller signal or control it using a side-chain input. There are also four 'direction' modes, offering different forward and backward repeat options that include reverse playback and Bounce, which alternates forward and reverse playback. The delay can also be bypassed to leave the limiters active to add their coloration when you don't need other effects. These add-ons appear in an extension panel populated with buttons and three knobs. The reference knob sets the reference signal level for the saturator, limiters and converters.

On the main panel, you'll find the expected delay controls for level, time, wet/dry mix and feedback plus a DLYx2 button that doubles the delay time by halving the sample rate, delivering a somewhat lo-fi sound character. Output gain is also provided, along with metering. High-cut filtering is controlled using a button. It defaults to a first-order filter at 6kHz, though this can be changed in the extension panel. FB Invert inverts the feedback signal, as you'd expect, while DLY Inv inverts the polarity of the wet signal. Switching occurs at zero-crossing points in the waveform, to reduce clicks as the switch is operated. When bypassed, the dry signal level is as set by the Mix knob. The mod section is pretty self-explanatory, with a single control balancing the square, sine and envelope mod contributions, and there's a phase control for offsetting the start of the LFO waveform. A bypass button on the lower panel does bypass the delay, but leaves the dry signal processed via an input limiter whenever Lims or Full processing mode is selected.

PSP42 v.2 can also be used as a phrase sampler/looper, and the display shows the current delay time in ms or bar/beat values. RPT mode sets the recording length by time, whereas Time mode works in bars and beats, with several subdivision options available. The loop speed and pitch can be changed using the x2 button or by adding modulation. Sonically the switches for bringing in the various coloration result in audible but still subtle changes to an otherwise clean signal.

Impressions

The charm of PSP42 has always lain in both its sound and its functionality, and both are improved in this updated version, while nothing appears to have been lost. As a delay/echo, you can get any of the expected single-head tape-echo treatments, and the modulation adds a little welcome movement. Reverse does the usual backwards echo trick but Bounce is an unexpected pleasure, adding an ethereal tail to whatever you throw its way. When used as a phrase looper or very long delay, the extended time of 28.8 seconds is welcome — anybody who fancies a dabble in Frippertronics will have a lot of fun. In short then, this is a very worthwhile update to PSP's original take on the Lexicon PCM42.

summary

A worthwhile update to a classic plug-in modelling classic hardware — it both sounds better and boasts greater functionality than its predecessor.

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Cinematique Instruments Marble 2

Virtual Instrument

Cinematique Instruments' inspirational VI gets a major reworking.

JOHN WALDEN

I always look forward to exploring a new Cinematique Instruments release because, more often than not, I know I'm going to get something that is just a little bit different from the mainstream virtual instrument crowd. In that respect, Marble 2 — yes, a follow-up to the original Marble instrument reviewed in the April 2016 issue of *SOS* — doesn't disappoint. It's a very attractive combination of familiar concepts, a somewhat quirky UI, and some unusual instrument choices and combinations.

What's New?

The underlying idea remains the same as the original. You get a carefully curated suite of 63 all-new core sound sources (each with round-robin and dynamic layers as required) that can be blended via the engine's two independent sound slots. Both of these slots offer a fully featured step sequencer with options to sequence 11 different parameters, including a range of sound-shaping/effects parameters. Compared to the original, the UI has had some restyling but there are also new features, the highlights of which are improved hands-on mod wheel sound design (via the Shape page) and a more efficient browser based upon 23 description filters and that improves the preset selection process.

The core sounds fall into four categories: organic sounds, powerful synths, textures and percussive instruments. While 60+ instruments might not sound like a lot compared to some modern sample libraries, as ably demonstrated by the 400 supplied presets, the number of potential pairings and the sound-design options provided mean that you can conjure a huge range of very usable sounds from Marble 2. Indeed, with the combination of excellent melodic sounds (both lead and bass), some impressive sustained/texture sounds and plenty of percussive/rhythmic options, you have all the sonic building blocks required to build a complete musical score.

Engine & Shape

The Engine page provides access to both the 16-step sequencer and the capability to browse for individual core sounds. You can specify different step counts for each of the two sound slots and then easily build note and sound parameter sequences using a familiar step-style grid system. Sound parameter sequences can actually be created using up to 64 steps and, via the Rep (event repetition) parameter, even with just 16 steps in the main note sequencer lane, you can generate some complex note/sound rhythms.

While the Engine page allows you to sequence changes in your sound, the Shape page allows you to map multiple parameters to the mod wheel for hands-on control in real time. This is neatly implemented and, essentially, allows you to create custom curves for the mapping of each available parameter through the full travel on the mod wheel. Obvious candidates are volume, note length and low-pass filter for each sound slot, but you can also control attack, sustain and drive, among some other options. The Random option is particularly interesting as this allows you to control a randomisation amount applied to volume, pan, tune or distortion for each sound source. The results can be somewhat unpredictable (I guess that's the point!) and a little goes a long way, but it's very cool in use.

Conclusion

While there are plenty of sound options available here, both in terms of the underlying sounds and the UI's options for shaping them, Marble 2 does have a particular sonic character. The blend of organic melodic, texture and percussive sounds, alongside the selection of (mostly) analogue-style synth sounds, gives Marble a suitably quirky vibe. As such, I'm not sure it's the kind of instrument you might pin a rom-com or super-hero score around. However, if your project suits a more experimental approach to its music, Marble 2 might be just the thing. Equally, if you need something a little different for your next crime-based/tension/drama cue, this instrument has plenty to offer. The price point is perhaps a little beyond 'causal purchase' territory but working media composers would find plenty to enjoy here and an engine that encourages the creative process. Hats off to Cinematique Instruments for keeping things left-of-centre in a world that overflowing with mainstream 'me too!' sample libraries; Marble 2 sounds great.

summary

Marble 2 is typical Cinematique Instruments; quirky, creative and slightly outside the mainstream. Ideal for media composers with an experimental streak.

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Wes Audio Phoebe

500-series Preamp

With the release of their 'Next Generation' preamp, Wes Audio have made it possible to assemble a full 500-series analogue channel strip with DAW plug-in remote control!

BOB THOMAS

In recent years, the dream of seamless hybrid analogue-digital workflows has driven a small number of manufacturers to develop analogue audio hardware that is, or can be, controlled by dedicated DAW plug-ins. Wes Audio are pioneers in this area, and have developed quite a wide range of digitally-controlled analogue (DCA) equalisers and compressors, in both the 500-series modular and 19-inch rackmount formats.

The new Phoebe, a single-slot 500-series module, is a mic/line/instrument preamplifier, and its addition to their NG (Next Generation) range means it's now possible to create a full Wes Audio 500-series DCA recording channel, complete with preamp and your choice of EQ and compression types. As with Wes Audio's other NG-series products, the Phoebe can be controlled by a proprietary DAW plug-in that communicates bi-directionally with the hardware over Wes Audio's GCon communications framework.

Hardware Overview

The Phoebe is a single-width module that carries Wes Audio's signature NG500-series white-painted fascia. Its restrained black and white cosmetics and elegant miniature LEDs convey an air of austere authority, but for the point size of its front-panel lettering, which would not be out of place on the bottom of an eye-test chart! The plug-in control might be appealing, but it's not necessary: the hardware can be used without hooking it up to a computer, so I'll cover this side of things first.

As with most microphone preamps, the Phoebe's user controls aren't exactly complex. The gain control is a continuous, detented push-and-turn encoder, with a circle of coloured LEDs and scaling indicating the gain being applied to mic (+20-75 dB, red LEDs) and line-level (-10 to +20 dB, green) sources. The level of instrument gain isn't shown on this scale, but is specified at -30 to +25 dB. Pressing the gain control switches the input, via a relay, between mic and line levels. This is followed by a vertical stack of four momentary buttons with LED indicators. The first three of these toggle, again with relays, 48V phantom power, polarity inversion and input impedance (1.2kΩ or 300Ω). The Phoebe can route, using the fourth button in the stack, one of two physical inputs to its transformer-balanced input stage: the rear-panel connector of the host 500-series rack or the balanced XLR/TRS combi jack on the module's front panel. Two of the button's three LEDs indicate which is in use. When the front-panel input is active, inserting a TS jack automatically switches it to high-impedance operation, illuminates the third LED to indicate that status, and switches off 48V phantom power.

The Phoebe's nine-LED output meter is scaled from -10 to +24 dB, and sits opposite the switch stack. Its input is taken after the output transformer but before Wes Audio's proprietary passive Iron Pad, whose continuous, detented control sits just below and between the meter and switch stack. Turning the control knob of this pad, which is always in circuit, activates a relay cascade that sets the level of attenuation (0dB to -15dB in 1dB steps) and illuminates the corresponding LEDs.

Sitting below the combi input is a mini-USB connector (with two LEDs) that allows DAW plug-in control of the Phoebe. The first LED (Data) indicates when communication is active, and the second (H-Link) confirms connection to the host DAW computer. Incidentally, Wes also offer a dedicated 10-slot 500-series chassis, the Titan, which allows communication with any »

Songwriter?

Turn to page 76.

» NG500 modules via a single rear-panel USB or Ethernet connection.

Circuit Highlights

The audio circuitry is fully analogue Class A, with a transformer-balanced, twin-amplifier topology that takes its inspiration, as others have before, from the work of the late Rupert Neve. The Carnhill input and output transformers are based on those used in classic Neve designs, and a cutout in the chassis metalwork creates just enough clearance for the chunky output transformer to fit within the confines of a single 500-series slot.

A clip detection function turns all the VU meter LEDs red when internal preamp distortion is detected. This makes it easy to maximise input-transformer-based harmonic distortion (by pushing the level of the input signal) while avoiding internal clipping (by judicious setting of the gain control). When you push the output transformer to persuade it to add its not inconsiderable weight and warmth to proceedings, you may well need to employ the Iron Pad to prevent the output from that transformer (which can reach a maximum of +28dBu) from overloading the next input stage in your signal chain. While it does give an indication of what's happening inside the preamp itself, a clip detector is a pretty blunt meter, and your ears will probably be a much better judge of the overall situation.

As with Wes' other NG500 modules, the hardware can optionally be controlled by a DAW plug-in — or the plug-in can be controlled by the hardware!

The lower input impedance setting is useful in allowing dynamic mics such as the YouTube pundit's favourite, the Shure SM7B, to give of their best, and is certainly essential when using passive ribbon microphones (especially vintage models). In my opinion, though, it's worth checking the effects of switching the input impedance for every mic in your collection, so you know both which value generally suits each mic best and how the 'less suitable' impedance changes its response — you never know when that knowledge will come in handy!

Before we move on to the plug-in side of the equation, it's worth noting that the Phoebe has two onboard, non-volatile memories. One (Memory A) stores the latest front-panel control status for the combi input and the other (Memory B) does the same for the rear XLR input. These memories are recalled simply by selecting the relevant input. This means that if you simply want to connect two separate sources, set them up and switch between them, you don't need to use the DAW plug-in — which is nice.

Softly Does It

Installing the DAW plug-in in Windows requires the installation of a proprietary driver beforehand, but no driver is required for macOS. The installation process is quick and simple, with firmware updated automatically when required, and user intervention only needed to select the models of GCon devices connected to the computer and to click on Next and Finish buttons. The installer also gives you access to further housekeeping and troubleshooting functions. Each version of the plug-in (VST 2, VST 3, AAX and AU) is installed in both mono and stereo versions — the latter eliminates the possibility of any difference between two Phoebes during stereo operation. Inserting the plug-in into a track brings up a GUI which displays all the physical controls and LEDs from Phoebe's front panel (in largely though not precisely the same layout) plus some extras.

It's possible to use a number of GCon devices simultaneously, so the first step with the plug-in is to assign an ID to the Phoebe. This is carried out by selecting the appropriate ID (only connected devices are displayed) from a drop-down menu at the bottom left of the plug-in GUI. The ID box also displays the source of the connection (USB if direct from a computer or a slot number if Phoebe is mounted in a Titan rack). Once an ID is assigned, bi-directional communication between plug-in and hardware is established, and the status and settings of all switches and controls synchronise, meaning a change on one results in an identical change on the other. The one functional enhancement that the plug-in brings to the Phoebe hardware is that the exact value in dB (to one decimal place) set by a control can be displayed on its virtual control when you touch its hardware equivalent.

Preset management is handled by a section at the top of the GUI. Each of the Front (A) and the Rear (B) Configuration Banks (Config) can store up to 20 preset configurations (Banks). It's a simple, intuitive setup that is fast and easy to operate.

Overall

Sonically, the Phoebe delivers everything you'd expect from a vintage-inspired, Class-A, Carnhill-transformer-equipped mic preamp. It wears its old-school, transformer-balanced low-end warmth, punchy mids and airy highs well, and when used standalone it's easy to work with. Its onboard front and rear status/setting memories are the icing on the cake — it would be a competitive product even without the DAW-based remote control, preset recall and automation.

The control plug-in, though, adds another dimension. It is completely stable, works exactly as designed and offers real convenience and flexibility to the professional and project engineer and the home recordist, whether used simply to store presets for individual mics or singer/instrument and mic pairings, to remotely control the gain and tonality during a live recording, or as a DAW-automated stereo pair for mixing through an analogue summing setup. If you like the British sound of the 1970s in a mic preamp, the idea of being able to store and recall presets, and/or if you want or need full automation or remote control, then the Wes Audio Phoebe NG500 preamplifier should be high up on your audition list. ■■■

summary

A great-sounding microphone preamplifier that delivers that quintessential 1970s British transformer-balanced sound, and whose plug-in-based remote control, preset recall and full automation offers flexibility, convenience and seamless integration with your DAW.

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McDSP KD1 Kinetic Drive

APB Saturation Plug-in

With the KD1 saturation effect, McDSP show that their analogue plug-in system can deliver more than just dynamics processing.

SAM INGLIS

McDSP's Analog Processing Box has been on the market for more than three years now, and is building up a devoted following among mix engineers in the know. You can read our November 2019 review of the original APB16 (www.soundonsound.com/reviews/mcdsp-apb-16) for a full overview, but in summary, it's a fiendishly clever device that takes the concept of digitally controlled analogue to its logical extreme.

The eight-channel APB8 or 16-channel APB16 connect to a Mac computer using Thunderbolt, and make their processing available through normal-looking plug-in interfaces. This gives the DAW user all the benefits of session recall and plug-in automation, but the processing itself takes place in the analogue domain. Signal routed to an APB plug-in gets sent out through a D-A converter before passing through an array of analogue components, with the various APB plug-ins configuring the analogue signal path in different ways so as to implement different processing characteristics. All of these are primarily based around manipulating the amplitude of the signal.

Super Saturation

The initial tranche of APB plug-ins were all compressors and limiters, but we've always known that the system is also capable of deliberately introducing distortion. Several of the existing plug-ins have saturation controls that allow the sometimes-desirable side-effects of overdriving analogue circuitry to be introduced. Until now, this has always been a secondary function, but with the new KD1 Kinetic Drive, McDSP have placed the APB's saturation capabilities centre stage.

Like all the other APB plug-ins so far, KD1 is free to registered owners of the hardware needed to run it, though you'll need an iLok account to authorise it. Each mono instance uses up one channel of APB processing. The interface presents six rotary controls and three radio buttons. The Drive, Wet level and Dry level controls should need no explanation, and the wet/dry balance can be locked so that you don't have to remember to reset it each time you load a preset.

The radio buttons are used to select your choice of three distortion circuits. As this is an analogue plug-in, that's not a metaphor: the circuitry really is reconfigured on the fly! The AWD circuit is

broadly neutral, adding harmonic content roughly evenly across all frequencies. By contrast, FWD has a fundamentally treble-centric sound, whilst RWD is darker. These are very general characteristics, perhaps akin in EQ terms to tilt or low-Q shelving bands.

Two of the remaining three rotary controls work in tandem with the choice of circuit to further shape the tone of the processed audio. These Focus frequency and Amount controls dial in a more specific form of emphasis, applying up to 12dB gain at your choice of frequency between 40Hz and 4kHz.

The most interesting feature, though, and the one that gives KD1 its name, is a rotary control labelled Kinetics. McDSP say that this "controls analogue-like behaviours, such as tube-like compression and voltage supply reaction". The Kinetics control runs from zero to 100 percent, and its effect is reflected visually on a bargraph meter labelled in Volts.

Light Touches

Some distortion effects major on extreme fuzz and dirt. Others explore the more subtle realms of light saturation and colour, and although terms like 'crush' crop up occasionally in the preset list, a little experiment suggests that KD1 is among the latter. If the source is recorded reasonably hot, high Drive and low Kinetics settings can produce outright distortion, but you're not going to get high-gain metal amp sounds from it. In fact, although the description might suggest that Kinetics models the 'sag' you get from valve rectifiers in guitar amps, KD1 definitely isn't a guitar amp modelling plug-in.

So what is it? Well, on one level, it's a very cool way of manipulating the tone of recorded audio. By balancing the choice of circuit with the settings of the Focus controls and the wet/dry balance, you can achieve a surprising range of results. For example, you might assume that the RWD circuit is the obvious choice for bass guitars or kick drums, and in a sense it is, but the Focus controls make the other circuits highly usable on sources with lots of low-frequency content. The broad treble emphasis applied by the FWD circuit can be paired with a more narrowly defined boost at 70 or 100 Hz applied using Focus and, at restrained settings of the Drive control, the result is not obvious distortion or saturation, but a subtle reshaping of the source that can make it sound more articulate and fit better into the mix. Or you could do the opposite, and use the RWD circuit to add heft to an anaemic kick drum whilst Focusing somewhere in the upper midrange to make it 'knock'.

The general principle of using the Focus control to complement the tonal shift applied by the choice of circuit works equally well with other sources, too. For example, you can turn KD1 into a very effective treble enhancer by Focusing high up the frequency range and blending in just a little of the wet signal. Struggling to restore some lower-midrange substance to a thin-sounding electric guitar, I tried pairing the AWD circuit with a Focus boost at 400Hz, and the sound instantly acquired the richness and warmth I'd hoped for, with no hint of mud. I don't think I could have achieved the same thing with EQ alone.

As the effect generally remains subtle across quite a wide range of control settings, KD1 is also eminently usable on groups and on the master bus. I've often thought that the APB system would be ideal for mastering engineers, who would love its precision and recallability, and this plug-in adds weight to that case.

Poetry In Motion

Digital emulation of analogue saturation has been getting better over the years, and as long as you left the Kinetics control at zero, you could perhaps get pretty close to the sound of KD1 using conventional plug-ins. But Kinetics really does do something I've not heard in software. Its most obvious effect is to back off the distortion, making things sound smoother, but its real magic lies in the complex dynamic response it introduces. The KD1 manual uses the term 'pumping' to describe this, but to my ears, it's more subtle and less straightforward than that expression suggests. Kinetics somehow introduces a sense of life and motion into almost any sound, but does so in such a way that you don't really notice it doing anything until you bypass it. Some say that digital distortion plug-ins tend to sound 'flat' and two-dimensional — and next to KD1 with its Kinetics control turned up, they certainly do.

"I've often thought that the APB system would be ideal for mastering engineers, who would love its precision and recallability."

Creating a successful hybrid product requires both analogue and digital expertise. You'd expect the latter from a company that has been developing plug-ins for several decades, but the APB system shows that McDSP are equally adept at analogue hardware design. If you're thinking that only the golden-eared will be able to tell the difference between analogue and digital in this day and age, a few hours spent auditioning these analogue plug-ins will very likely change your mind. The distinctive quality of this saturation effect only reinforces that perception.

summary

Simple to use yet surprisingly deep and uniquely rich-sounding, KD1 Kinetic Drive is another powerful demonstration of the value of McDSP's unique analogue plug-in platform.

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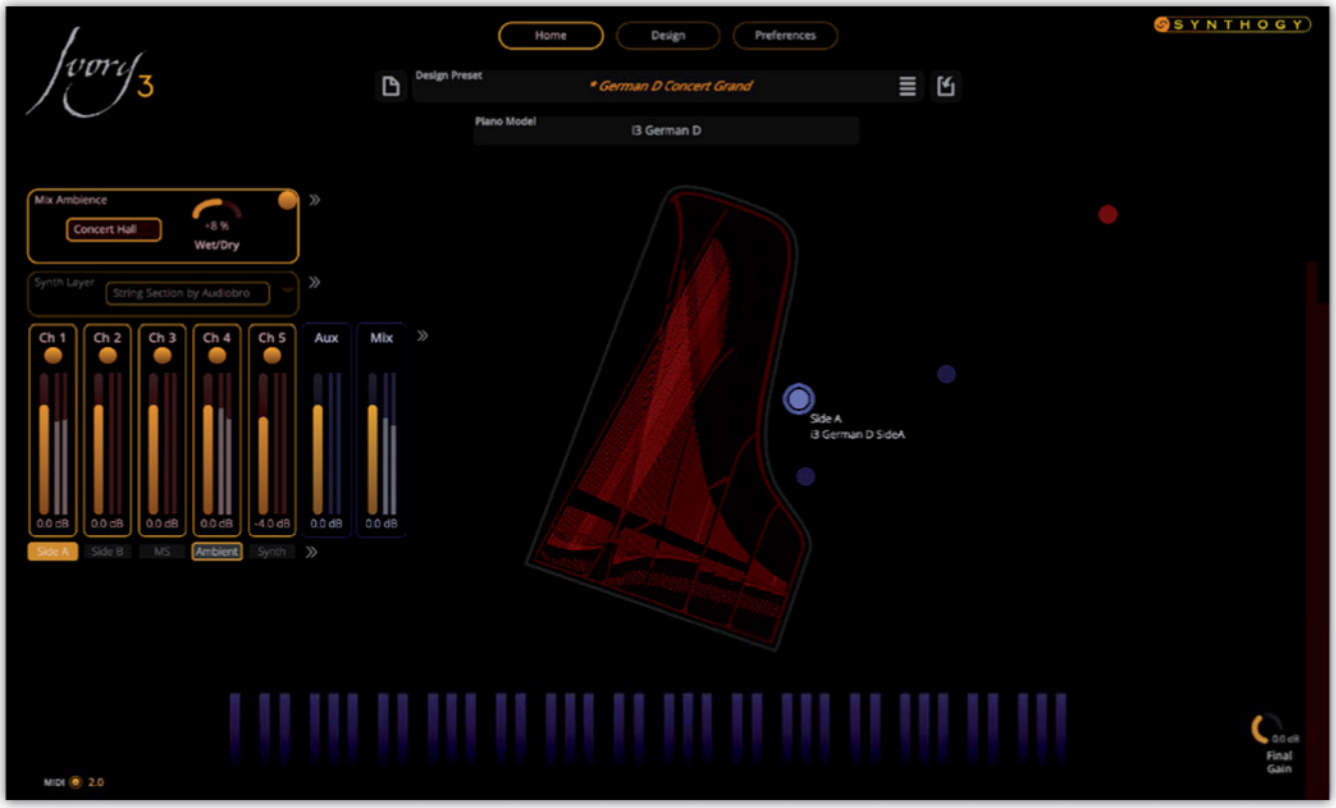

ROBIN BIGWOOD

Synthogy Ivory 3

Virtual Piano Instrument

Is Synthogy's Ivory 3 the best virtual piano you can buy?

Synthogy's Ivory has been around a long time, in sample library terms. Its roots go all the way back to at least 2005, and I reviewed Ivory II Grand Pianos in 2011. Along the way there have been various side hustles: individual Steinways, upright pianos and studio grands, using the same playback engine. It has always been among the major players, combining fine sound quality and playability with a surprising scope for creative sound design. I've personally used it on dozens of jobs.

Now Ivory 3 German D brings the technology up to date for the present day. It's still a self-contained plug-in (VST3, AU and AAX) and standalone application for macOS, with a Windows version in the pipeline. But this new flagship version of the software now places all its eggs in one piano (please don't try that at home): a single German-variety Steinway D-274, recorded in a concert hall in Quebec, and presenting your computer with an installation size of 42GB.

The New

Ivory 3's headline new feature is Continuous Velocity, powered by a Real-time Gradient Blending (RGB) engine. This promises smooth gradation of timbre in response to velocity, without any obvious sample switching. And not only for the traditional 128-value resolution of MIDI velocity, but for the cutting-edge spec of MIDI 2.0 too, with its 65,536 velocity increments, as well as the 16,384 of the MIDI Velocity Extension scheme, employed by a few controller manufacturers for many years now, which works by supplementing velocity with MIDI CC88 data.

The lack of obvious sample switching is by itself desirable of course, as it gets closer to the experience of playing a real piano, but it also brings Ivory 3 on par with pure modelling-based piano plug-ins like Pianoteq that have never been subject to the limitations of discrete samples. Synthogy are understandably protective of the details of the RGB tech, but it appears that it's still solely based on sample replay, probably with some filtering or other manipulation in the moment, as opposed to formally combining an acoustic modelling element. What's for sure is that Ivory 3's Steinway sample set does not present itself in the software the Ivory II pianos used to, which was in different versions with specific numbers of velocity layers, and some dark/soft or bright/loud skewed alternatives. Now there's just the one, and some new parameters (Dynamic Shift and Hammer Strength) in the user interface that can recreate those tilts, and more besides.

Another big enhancement is a multi-mic capture and mixing scheme. Four mic perspectives are on hand, and supporting them an onboard mixer that has a stereo width module, dynamics processing, a three-band EQ and an aux send on every channel. Two dedicated aux and master mix channels additionally get chorus and ambience/reverb processors. Two mic perspectives, called Side A and Side B, were apparently captured with small-diaphragm cardioid capacitor mics close to the piano. An M-S pair is, of course, a Middle and Sides array, placed at a medium distance. Finally, small-diaphragm »

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» omnis were the basis for an Ambient array that's positively loose and roomy.

What else? The user interface has been completely rebuilt, and its high-DPI tabbed and scrolling design looks and feels much more up to date than any previous version of Ivory. Also listed as a v3 benefit are enhanced sympathetic resonance and sustain resonance processing, along with some new synth layers, including an ensemble strings sample and an FM piano. There's a simplified preset structure too, replacing the previous potentially confusing scheme of Programs, Sessions and Effects.

The Familiar

At the same time, it's perhaps reassuring to note how similar Ivory 3 is to previous versions. So much so, in fact, that it provides full backwards compatibility with their sample sets, and those old preset types.

Parameters like dynamic range are still there and remain hugely valuable: lowering the range boosts the playback level of lower velocity notes, making for tones that are both delicate and yet loud and present. Something like what a compressor would do, but without the sucking and pumping.

Similarly, Ivory's sustain/damper pedal implementation is one of the best in the business. It supports, very well, half-pedalling (quick or shallow pedal releases that don't fully damp pedal-held notes) and partial pedalling. You need a continuous-type damper pedal to get the best from it, and then you can set the points in the pedal's travel at which different behaviours occur. It can work well with simple switches too.

Velocity handling is another strong point. If your controller won't readily spit out the highest velocity values, or does so too readily, Ivory can compensate. Input/output velocity mapping is shown on a graph-like display which can be manipulated via accompanying controls. It works very effectively.

There are good options for controlling pitch and tuning too. Not only the base pitch of A4 (which defaults to 440Hz) but transposition in octaves, semitones and cents. A Stretch Tuning option recreates what good piano tuners do, widening octaves towards the extremes of pitch to compensate for string inharmonicity. A good handful of temperaments is provided too, and Ivory will also read tuning info from your DAW, if it's capable of providing it.

In Practice

So now to the really important bit: what is Ivory 3 like to actually live and work with? Simply, I think it sounds and feels absolutely superb. To my ear the new Hamburg Steinway is a great piano sound, oozing class, and very versatile: it seems equally well suited to pop, jazz and classical genres. The capture quality of the Side A mic array in particular is stunning, almost three-dimensional, and phenomenally full and detailed. Note decays extend very naturally — not much short of 50 seconds for the lowest bass notes — and there's equal focus and immediacy in the capture across the entire pitch range.

During the review period I was fortunate enough to do several days' playing work with a well-maintained Steinway D, in a good acoustic, and it's to Synthogy's credit that they've captured so much that is appealing about the real thing, in all its six-figure cost, 500Kg glory. For one thing, the way that these big beasts are so controllable, delicate one moment and then so singing, commanding or even thunderous the next, and with a vast range of character ready to be teased out with just small changes in touch. Synthogy have got this dynamic response aspect spot on.

Comparing a representative cross-section of other good virtual pianos is interesting, and helps pinpoint individual qualities of Ivory 3. Garritan's CFX Concert Grand, now almost a decade old, is similarly classy, and sounds more light, bright and glossy, without the inherent 'woodiness' of the Steinway D in a few places across the pitch range. However, Ivory 3 has a far more present and willing treble register, somehow more gravitas »

Will It Blend?

Aside from all the normal testing I undertook during my time with Ivory 3 — playing normally, listening to every note individually, assessing velocity response, exploring parameters — I wanted to dig into the new Continuous Velocity and Real-time Gradient Blending feature.

To do that I set up a project in my DAW (Studio One) that played several different notes hundreds of times, with a smooth, linear velocity gradation generated across them with an editing tool. PreSonus do not claim high resolution or MIDI 2.0 compatibility for Studio One, but at the very least this test would generate note triggers at all 128 values of the 'original' MIDI spec. Certainly enough to hear what the new RGB engine is doing. (As an aside, Ivory 3 does flash up a MIDI 2.0 label when it receives data from Studio One, so maybe there's something PreSonus are not telling us...)

In short, there was good, smooth gradation of tone and no big lurches in timbre as this MIDI test played back: exactly what the RGB engine promises. Brightness and dynamics increase gradually and naturally, just as you'd hope. There are still some audible timbre shifts — yes, almost certainly velocity layer switches — but they're subtle.

By way of comparison, I played the same MIDI into Ivory II. That is much older software of course, which was always up front about its (at most) 18 discrete velocity layers; although interestingly Ivory II also claimed a 'Sample Interpolation Technology' for smooth transitions, in its day. Here, velocity transitions were clearly, glaringly audible. And in fact something similar can be recreated in Ivory 3 by disabling the RGB engine. I find it interesting this is even offered as a user option, but there we are.

Is this a good metric by which to judge Ivory 3? Yes and no. No, in that this sort of testing is horribly, forensically, pixel-peepingly amusical, and has very little to do with most musicians' playing experience. But yes in the sense that the Continuous Velocity feature does indeed stand up to scrutiny, and smoother transitions, being closer to acoustic piano behaviour, can only be a good thing.

Synthogy Ivory 3
$279

PROS
- A particularly classy-sounding Steinway D captured with glorious detail and focus.
- Continuous Velocity does away with obvious velocity-driven sample switching, and equips Ivory for the MIDI 2.0 future.
- Onboard multi-mic mixing gives new options for piano character, and more flexibility in effects processing than ever before.
- Useful improvements to many existing strong Ivory II features, but with full backwards compatibility for libraries and presets.

CONS
- Distant mic perspectives are implemented in a way that can make them sound curiously dry.

SUMMARY
A brilliant-sounding, jaw-droppingly responsive Steinway D, supplemented with a broad range of sound-design options via onboard mixing, EQ and effects, and sample manipulation. A rock-solid option as your go-to piano for all sorts of jobs.

zynaptiq

science, not fiction

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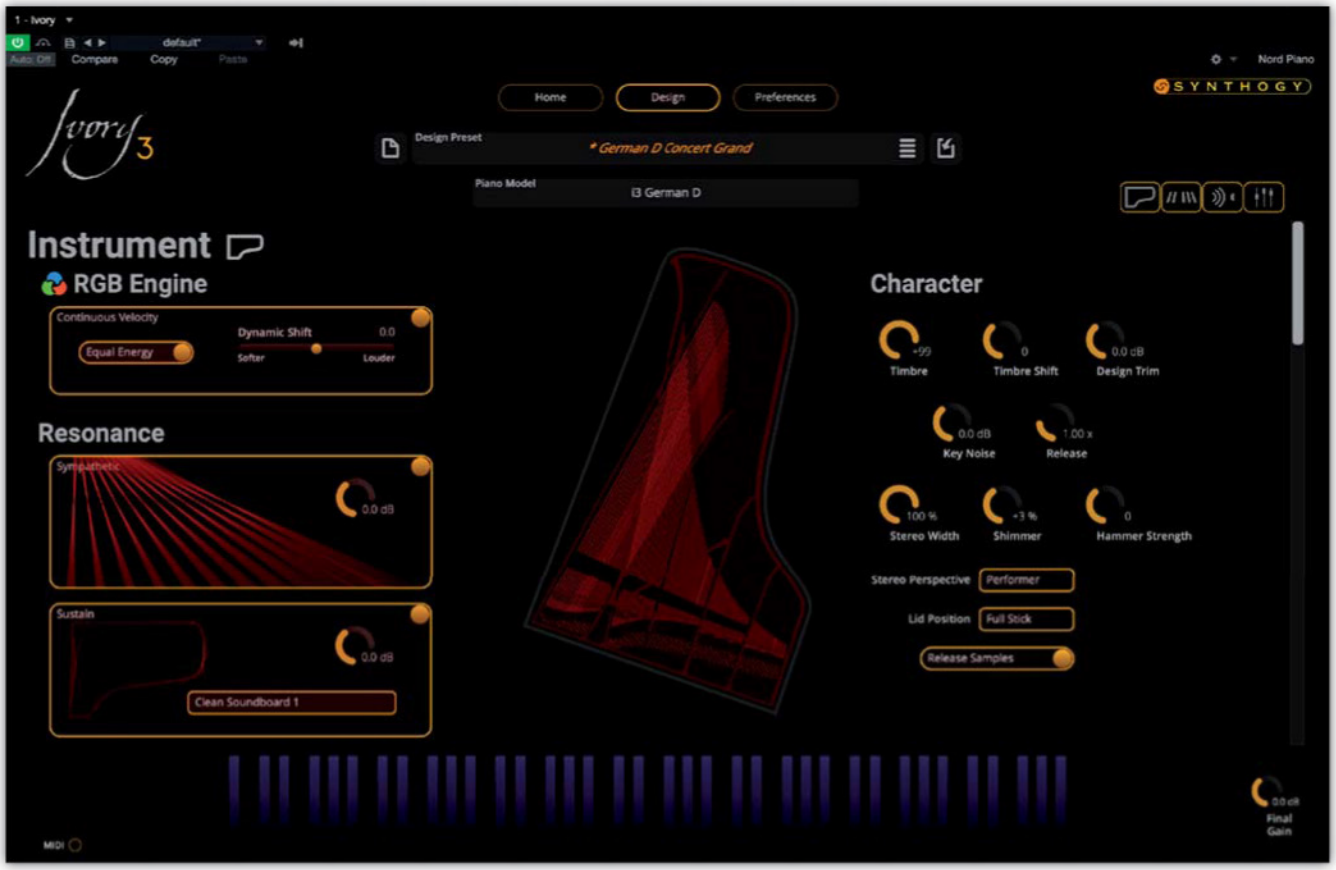

Ivory has some interesting scope for sound design, and version 3 builds on what was already there. Naturalistic tweaks like lid position and sympathetic resonance levels can be made. But Timbre, Timbre Shift, Shimmer and Hammer Strength parameters can change the sound both subtly and wildly. Together with the onboard reverb options they're especially good for a range of desirable, quasi-cinematic saturated, spacious, sensuous effects.

» overall, and a much greater scope for tone tweaks.

Just because I was using a Nord Piano 3 as my controller, I couldn't help comparing some of its pianos, from the hardware-only Nord library. These have no right to sound as good as they do for their comparatively tiny file sizes, and several share Ivory 3's sense of holographic presence. The sheer depth of Ivory's sampling though, and those smooth velocity transitions, win out under close examination.

Finally, Pianoteq 8's German Steinway D acquits itself remarkably well. This acoustic modelled alternative gives an extremely involving playing experience, and it shares with Ivory 3 a tremendous sense of immediacy and focus. I suspect some may judge it to be a little less subjectively attractive though.

Ivory 3's mic perspectives other than Side A offer quite different characters. Side B is more focused, serious, borderline constrained in tone, still on the dry side, but with both a bit more room and 'fibre' in the sound. It's useful. The M-S pair is more distant, and perhaps lends itself best to a classical-type sound, for blending with an orchestral sample library. It will also, of course, give a true mono feed, with no phase or blending issues, via the mixer's M-S processing. The ambient mic feed, meanwhile, has an even more spacious, open, omni-like quality, less focused but pleasantly large.

The mic mixing feature is good to see. It has some unexpected features too, like being able to direct a single mic feed to more than one channel in the mixer, allowing for parallel compression techniques, for example. Also time-alignment between arrays can be manually adjusted, and polarities inverted. But there are aspects that could be better.

For example, the mixer channels and the mic arrays that feed them are never simultaneously visible in the graphic interface. You can scroll backwards and forwards but I'd have much preferred to have sources, signal routing and mixing all in one single screen. Or at least a way for mixer channels to be named by what's feeding them, or named manually.

Then, my only real gripe about Ivory 3 is the way the way the M-S and Ambient mics incorporate the sound of the acoustic in which the piano was recorded. Or rather, don't! These more spacious captures include a good amount of the contribution of the acoustic which is audible 'around' held notes at any dynamic level. But when any that were played with less than about 60 percent velocity are released, the expected acoustic decay doesn't happen. It's as if the piano had been recorded in a completely dry acoustic. Louder notes do, suddenly, incorporate a bit of reverberant tail, but it's at a surprisingly low level.

To my mind this behaviour is odd, because it means you're bound to still use one of the onboard Ambience processors (or a reverb in your DAW) to get something that sounds natural and plausible even when you're using only the spacious mics. And then you end up with a double acoustic: one natural as captured in the sample, and one artificial.

Synthogy told me that this was intended, and pointed to the effectiveness of many of their 'Cinematic-Ambient' presets, which use the Ambient mics and reverb together. I can't argue with that: »

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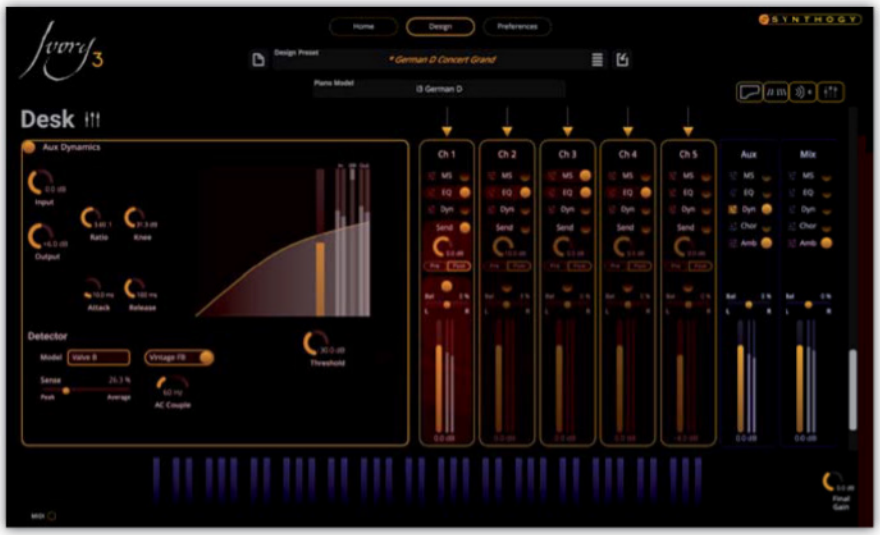
» many sound excellent. At the same time, I wonder if they might consider adding a release sample level control alongside the existing release time multiplier. That'd let you work with captured ambience, artificial reverb, or both, without breaking any backwards compatibility.

I've few other criticisms. Ivory 3 can be quite CPU-hungry, especially with several mic arrays enabled, but while Studio One's CPU meter lit up with enthusiastic playing using a brisk 64-sample buffer size, my M1 MacBook Pro shrugged it off in fanless silence, with lots more happening simultaneously, and barely a murmur in the OS's Activity Monitor app. It should be a similar picture with other modern CPUs. The FM piano layer is also curious. Useful in some circumstances certainly, but with positively clunky transitions between just three velocity layers, which only stand out the more next to the Continuous Velocity Steinway.

Countering these criticisms, and to set the balance straight, are many very strong positives. That fundamental sound quality and responsiveness I've already mentioned: really, really impressive, and addictive. Adding to this, the already significant and powerful sound-design possibilities are enhanced further by the new Hammer Strength parameter. At one extreme this takes the Steinway D into dreamy felt-piano territory: soft, fuzzy, diffuse. At the other, tight, snarling rock and dance piano timbres. Combine it with Timbre Shift and other pre-existing parameters and the pristine D-274 starting point piano can be twisted out of all recognition, and nearly always into something useful, interesting and engaging. Meanwhile the sustain and sympathetic resonance features continue to add a great deal to the sense of realism and complexity. Amongst other things they let you adjust the damper pedal's sound from naturalistic, adding breadth and a sense of scale, to something much more liquid, luscious and immersive.

In the mixer, the Dynamics sections are surprisingly fully-featured. Their different detector models allow a range of subtle to more overtly pumping and noisy compression effects to be dialled in. EQ is simple — three-band, with low and high shelves and a parametric mid — but effective.

The third and fourth pages of the Design tab show available sound sources and a routing matrix, which feed a mixer below. The need to scroll to see what's connected to what, and the generic numbered mixer channels, I found a little irksome. Onboard effects processing is quite fully featured though, and parameters for the different modules appear to the left of the mixer.



Conclusion

Ivory 3 is a really worthwhile update. All previous Ivory strengths are left intact, but are now combined with the best single piano capture Synthogy have ever offered, and about the most sophisticated velocity response of any sampled piano out there.

What I've always enjoyed about Ivory — whether jamming with it, or relying on it to meet a production deadline — is the way it gets the fundamental sound and playability right from word go, and adapts unfussily to different controllers, DAWs and workflows. But it also supports beautiful-sounding creative journeys when you want that. Which is to say it can do respectable and natural, sumptuous and cinematic, as well as beautifully weird. Ivory 3 takes all that up a gear.

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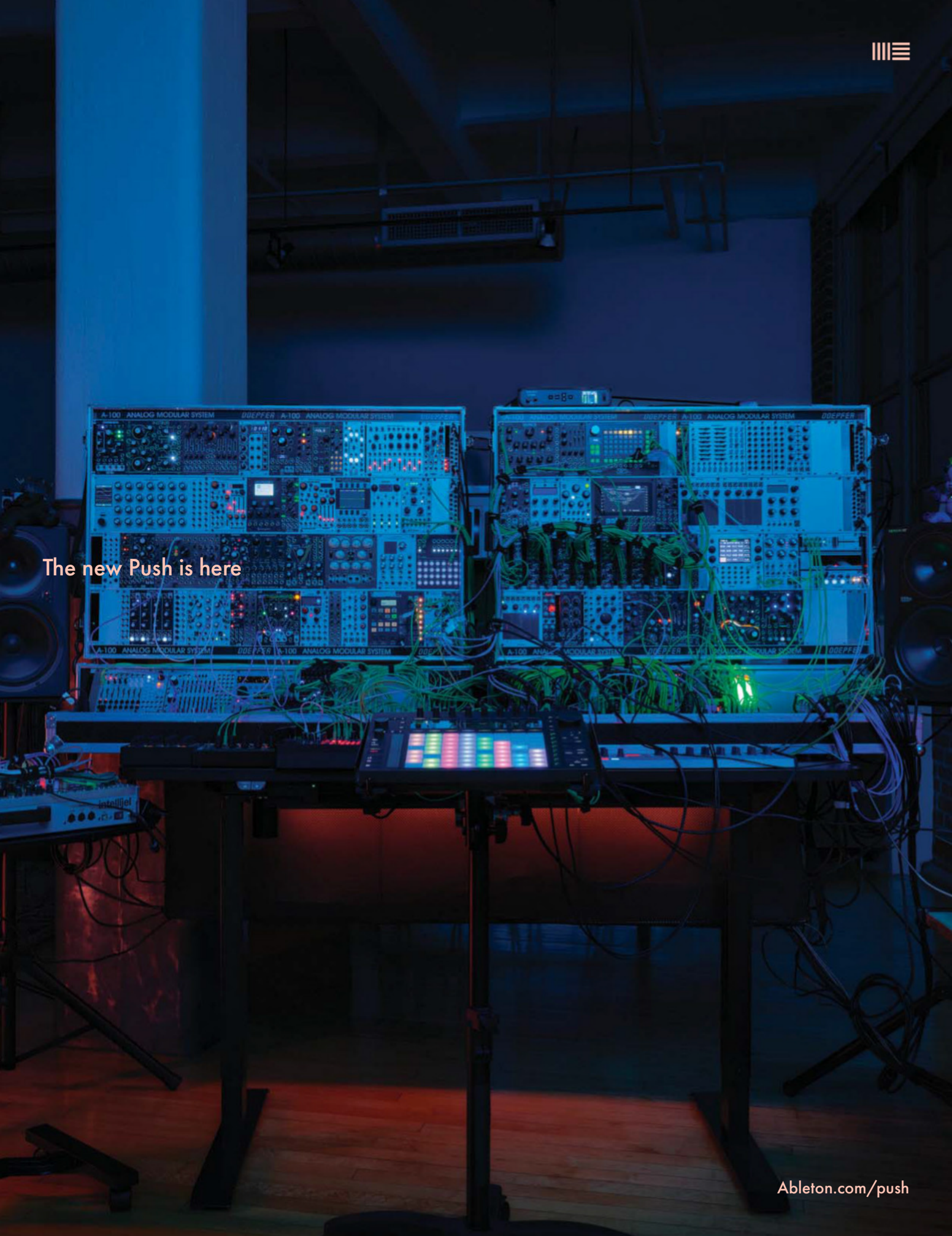
System Requirements

You need quite a good-spec computer to run Ivory 3: for the currently available macOS version Synthogy say an eight-core Intel chip at minimum, and an Apple Silicon CPU recommended, and you'll need to be running macOS 10.15 or newer. Similarly, it's 16GB RAM minimum, 32GB recommended, and an SSD is *de rigeur*.

Copy protection and authorisation is via iLok, but that can be a virtual authorisation on your hard drive, or in the iLok Cloud, and a physical dongle is only optional. You'll need an iLok account, but after that, authorisation is a painless process. This scheme must surely be amongst the most flexible out there in letting users tailor the boring reality of copy protection to their own workflows.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PROFESSIONAL AUDIO HARDWARE AND SOFTWARE

Lake People MC100

Monitor Controller

The monitor controller is the hub of the 21st Century studio, and if you have the budget, Lake People's MC100 has all the I/O you'll ever need.

SAM INGLIS

As I grow old and boring, I find myself more appreciative of mundane studio gear. Maybe I've just been doing this job too long, but I can now become genuinely enthused by orchestral mic stands, or clever shockmounts, or a well-organised patchbay. And if you'd told the younger me that his older self would one day wax lyrical about monitor controllers, I suspect the younger me would have had some choice words to say in reply.

It's a good thing the younger me isn't around to express contempt, then, because the 2023 version removed the Lake People MC100 from its box with a surprisingly keen sense of anticipation. That's partly because hands-on monitor control strikes me as something that's increasingly important in modern studios, where there's rarely a console master section to handle volume control, speaker switching and so on. And it's partly because the MC100 is the very epitome of a functional, engineering-led design. This is a piece of gear designed to do a job, not to look cool or impress clients.

Not that the German-made MC100 looks bad. It occupies a surprisingly hefty metal case about 2U high, and is finished all over in matte black. Legending is clear and precise, LEDs are bright, and the action of all the controls is confident and positive. These include no fewer than 19 buttons, plus two potentiometers relating to the headphone output, and the huge 50mm main volume control. The MC100 is actually available in two versions, which differ only in the operation of this control. The standard version, which was supplied for review, uses an ALPS RK27 potentiometer with 41 detented positions, whilst the MC100 Pro uses a relay-controlled attenuator with 256

steps. The MC100 is an active monitor controller and draws its power from an external 'line lump' PSU.

No Jack City

The MC100 has a total of five stereo input paths, while on the output side, it supports three sets of stereo speakers, with an additional subwoofer output and headphone out. In a fit of purism, Lake People have implemented all the balanced I/O on XLRs only, with nary a quarter-inch socket to be seen apart from the headphone connector. This jack-free zone comprises the first two stereo inputs, the three main speaker outputs, the mono subwoofer output and two pairs of 'slave' outputs, while the full input quota is completed by a pair of RCA phonos, a front-panel mini-jack and a stereo USB input. There may be sound engineering reasons for preferring XLRs to jacks or combi sockets, but let's face it, if you were going to apply that logic consistently, you wouldn't include phonos or mini-jacks either, so this might seem a touch wilful on Lake People's part.

Each input and output has its own dedicated and clearly labelled button, and the MC100's inputs and outputs can be toggled independently between exclusive and additive modes. In the exclusive mode, which is the default, selecting a new input or output automatically de-selects the existing one. In additive mode, you can have as many active at once as you like, and the contributions of multiple inputs are summed. The subwoofer output can be toggled on and off independently whether or not the outputs are in additive mode, as you'd expect. There's no bass management, so all of the outputs, including the sub, always deliver a full-range signal. A welcome side-effect of this is that the sub output could be used to connect a single mono loudspeaker such as an Auratone. The 'slave' outputs always pass the input signal at full level, unaffected by any of the other controls, so are ideal for connecting hardware meters, or for those who like to have a stereo recorder running full-time to ensure that musical ideas are never lost.

One of the more unusual features of Lake People's G-series headphone amps (www.soundonsound.com/reviews/lake-people-g103-mkii-g105-mkii) is something they call 'pre-gain': fixed gain or attenuation blocks that can be switched in and out of the signal path to make sure that the variable controls, such as headphone or speaker level, are useful over as wide a range as possible. These are present and correct on the MC100's headphone amp, where they're set using internal DIP switches. But there's also pre- and post-gain available in various other places. All three of the speaker output pairs, and the subwoofer output, have rear-panel switches that introduce either 6 or 12 dB of attenuation, while the RCA and mini-jack inputs can be switched between +4dBu and -10dBV operating levels. Once again, I can't fault the engineering argument for having this sort of setting be switched rather than continuous, but if you need to match the subjective level across two different speaker systems, you're likely to need a finer degree of trim than is available here.

Swap Shop

Input and output selection accounts for barely half of the MC100's quota of buttons. There's also a front-panel power button, which is close enough to the headphone socket that you could accidentally press it whilst plugging in a pair of cans. Power up and down are pleasingly silent, with no thumps or clicks. There are Mute, Dim and Mono buttons, arranged in a broad arc around the main level control. And there are buttons to solo the left and right inputs, swap over the channels, and reverse the polarity of the right channel. (On the review unit, engaging Mute disables all the other buttons. I understand this is a firmware bug and will be fixed soon.)

These controls all do what they say on the tin, and are genuinely useful, but there are a couple of ways in which they don't align with my own preferences. First of all, although the Mute button does not affect the headphone feed, the Dim and Mono buttons do. In an ideal world, I think that would be optional. More of an issue for me is that the Solo buttons operate upstream of the Mono button. What I want to happen when »

Lake People MC100
€1799

PROS
- Comprehensive I/O selection, with the option of exclusive or additive source and destination switching.
- Very clean sound and perfect tracking throughout the volume range.
- Detented control makes volume settings easily recallable.
- USB input adds considerably to its versatility and supports DSD.
- Additional 'slave' outputs are handy for metering and so on.
- High-quality headphone amp with neat crossfeed feature.

CONS
- No quarter-inch I/O.
- The Solo buttons would be more useful if they followed the Mono button rather than preceding it.
- No talkback.
- The price will put some people off.

SUMMARY

A monitor controller of uncompromising design and build quality, which drips with thoughtful features.

The MC100's rear panel is a jack-free zone!

» I press Solo L and Mono at the same time is for a mono sum of both channels to appear only on the left speaker output. What actually happens is that the soloed left input channel appears on both speaker outputs. The ability to audition a mix in mono over a single speaker is very useful: the ability to audition one channel on both speakers, less so.

Standalone monitor controllers derive ultimately from mixing console master sections, and consequently, some of them include talkback. The MC100 doesn't, though you could probably improvise something using the front-panel mini-jack input if you didn't mind it always operating in latching mode.

USB & Them

The MC100 signal path is, for the most part, all-analogue, but the fifth input is on a USB Type-C socket. Intriguingly, this not only supports stereo PCM audio at sample rates of up to 384kHz, but can also carry DSD audio, the single-bit format that still has its fans in some audiophile quarters. More prosaically, it can be used to connect a phone or other portable device, and I had no problem selecting it as the system output for my Mac whilst simultaneously running DAW output through my audio interface. This is one of those features that's not glamorous, but certainly makes life easier.

I've already mentioned the headphone amplifier, which is similar in design to Lake People's standalone G-series models and sounds equally good. What this offers that those devices don't is crossfeed: the ability to bleed some of the left input into the right channel and *vice versa*. This is a long-established and effective way of making headphone listening sound somewhat more speaker-like, and the implementation here is excellent. The subjective level remains exactly the same at all settings of the X-Feed pot, and there's a separate button that toggles it on and off.

"Channel matching at low levels is usually the Achilles' heel of volume controls based around dual-gang potentiometers, but the review MC100 tracked perfectly down to the lowest setting, with no hint of an offset to either side."

On paper, the relay-switched attenuation in the MC100 Pro should offer better crosstalk, channel matching and longevity compared with the standard version. I didn't have the opportunity to compare, but I can say with some confidence that the standard version is easily good enough for me! Channel matching at low levels is usually the Achilles' heel of volume controls based around dual-gang potentiometers, but the review MC100 tracked perfectly down to the lowest setting, with no hint of an offset to either side. The 41 detented steps provide fine enough level control that you can set exactly the volume you want, but are clearly enough differentiated that you can recall settings precisely, for example if you want to work to a calibrated speaker level.

In terms of features, apart from talkback and its inability to provide a single-speaker mono feed apart from the subwoofer output, the MC100 does everything I could want from a monitor controller. Three pairs of speakers is enough for most studios, and the 'slave' outputs mean you don't have to sacrifice an output pair for metering or for capture to a stereo recorder. On the input front, all traditional bases are covered, and the USB socket greatly increases the MC100's flexibility. And of all the features that a monitor controller should have, the most important is that it must not colour the sound in any way. The MC100 sounds supremely transparent to my ears, exactly as it should.

So, having tried the MC100, I'm still enthusiastic about it! However, it must be admitted that Lake People's uncompromising design approach, and their commitment to manufacturing in Germany, make for a substantial price tag. The MC100's most obvious rival, for example, is SPL's MTC MkII; and despite costing considerably less, the MTC has built-in talkback and offers two headphone outputs to the MC100's one, both of which feature a similar crossfeed system. Most of its I/O is also on quarter-inch jacks rather than XLRs. But it lacks the MC100's highly versatile digital input and its front-panel mini-jack socket, as well as the Solo buttons and some of its gain adjustment toggles.

Can you buy something that has the same basic features as the MC100 for substantially less, then? Yes. But will you regret opting for something that's built like a tank and which bristles with nice additional touches? I doubt it. If you're the sort of person who believes in buying once and buying right, the MC100 is likely to make an enthusiast of you, too.

$ MC100 €1799; MC100 Pro €2299.
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CB Electronics XP-Relay

Analogue Routing Matrix

The XP-Relay combines CB Electronics' sophisticated patchbay control system with the purest possible signal path.

MATT HOUGHTON

Regular readers will know how impressed I was with CB Electronics' XPatch 32, which I reviewed in *SOS* August 2021 (www.soundonsound.com/reviews/cb-electronics-xpatch-32) — in fact, I stated that "if I were in the market for such a device right now, I'd be buying this one." The XPatch ecosystem has grown better since then, too. Not only have the promised higher channel-count versions become available, but the software control app is more intuitive and feature-rich. As with their main competition, notably Flock Audio's Patch, the XPatches employ active electronics and the routing is performed using amplifiers. There's nothing inherently wrong with using amps for the switching, particularly if you exploit their benefits has CB did on the XPatch devices (CB achieved commendable noise and distortion figures, and the amps make possible both very fine calibration of each channel for each attached device and, since my review, level automation) but for some people the idea of anything other than a 'straight wire' connection between devices is anathema, particularly if they have no need of the extra functionality that active electronics can deliver. Such people should find CB's latest product, the XP-Relay, rather more enticing...

Overview

As the name implies, the XP-Relay is closely related to the XPatches, but there are also fundamental differences in design and it joins rather than replaces them in CB's range. Like the XPatches, it is a digitally controlled analogue routing matrix, and it's controlled by the same XPatch app (which can communicate with up to four units of either type connected to the same Mac or Windows computer). But in this case the signal routing of the analogue signal is performed by relays, rather than amps, and the main benefit of this is that there is literally never anything more than a 'wire' connection between the inputs and outputs — just as would be the case on a traditional open patchbay. Also, the use of sealed relays means it should be less prone to contact problems than such bays, and unlike the XPatches it's possible to pass through phantom power from preamps — and since all 'pins' make contact only with those of their intended destination there aren't the same risks in handling phantom power as there are with a conventional jack patchbay. (As I'll cover in more detail below, the XP-Relay is itself also capable of supplying mics with +48V phantom power).

Of course, there are also some potential disadvantages compared with the XPatches. Since there's no amplification or buffering you can't apply gain offsets for different devices/channels, there's no possibility of level automation, and you can't, for example, route a guitar or bass DI to a line-level device without involving other hardware to take care of the different levels and impedances. Also, the maximum channel count for the XP-Relay is lower, since relays are bulkier and more expensive than amps, and other new facilities take up space on the inside and panels too. Currently, the maximum size of XP-Relay, which is the configuration sent for review (I'll discuss other options below), is a 16x16 matrix. It's possible, though, to use the control software to link two such devices for stereo use, one unit handling 16 left-channel signals and the other the corresponding 16 right-channel signals. In short, they're similar devices but not the same, and if you're contemplating a purchase you'll need to weigh the relative pros and cons according to your needs.

As with the XPatch, the real 'cleverness' of the XP-Relay, from a user point of view, lies in the XPatch control app. This has been updated frequently, and at the time of writing we'd reached XPatch version 4.4, which represents a significant improvement over v3.3 that I'd been evaluating when writing my XPatch 32 review. For one thing, it now supports drag-and-drop signal path configuration, which makes setup easier, but there are also plenty of smaller touches that combine to make it considerably more intuitive in general. You need to use the software to specify what gear is attached to what I/O and to set up your various routing options, but

The XP-Relay can communicate with its Mac/Windows control app over USB or Ethernet, and the latter allows connection to a router for , among other things, wireless MIDI control from a tablet or smartphone app such as TouchOSC.

thereafter use of the app to control the switching is optional. You can use the software alone if you wish, but there's now built-in support for switching using an Elgato Stream Deck (see box), and the hardware can also be operated standalone, with support for MIDI control and footswitch-based preset-scrolling. All of which opens up all sorts of possibilities in the studio and on stage.

Hardware

Extending a shade over 250mm behind the rack ears, the XP-Relay is pretty large for a 1U 19-inch rackmount device. The review unit was configured for 16x16 mono channels, and the other available options are mono 8x8, stereo 8x8 and mono 16x8 — and I'm told that after-sale reconfigurations are possible, which is nice. The internal PSU accepts mains AC power through a rear-panel IEC inlet and, being a linear rather than switch-mode supply, there's a rotational switch to select the AC voltage (115/230 V, 50-60 Hz). You need a screwdriver to change this, which I regard as a sensible precaution.

Also on the rear panel are the balanced analogue inputs and outputs, in the form of four DB25 D-sub connectors wired to the usual AES/Tascam standard. Alongside these are a USB B-type connector, a five-pin MIDI in DIN socket and an RJ45 socket. The XP-Relay can communicate with the XPatch app via USB or Ethernet — the latter allows a directly connected computer to be placed further from the XP-Relay, or, optionally, to connect the XP-Relay up to a WiFi-equipped router to allow wireless connection to a computer or tablet/smartphone running a third-party MIDI control app.

The front panel is decked out in CB's usual blue-on-bare-metal livery, and the controls and connections that it hosts are equally minimalist. On the left, a pair of XLR sockets duplicate inputs 1+2 on the rear. Working in from the right, a quarter-inch footswitch jack socket is followed by a knurled-metal push-turn encoder, a small but crisp colour OLED screen, and a power/data-present LED. In the middle, eight low-profile black buttons are arranged in a row, with an indicator LED above each one, hinting at functionality not present on the XPatches.

The XLR inputs will obviously be handy for musicians or engineers wanting to patch in different mics or line-level instruments quickly, but unlike the XPatches the XP-Relay has no outputs on the front. To some extent that's understandable, given the lower number of channels, the larger components and additional facilities inside, but it means it's not possible for, say, a mastering engineer or *SOS* reviewer to audition a new device without accessing the D-sub looms you've previously plumbed into the rear of the device and your existing gear; maybe this is something CB could consider addressing in a future MkII version.

The eight buttons were originally conceived to allow phantom power to be switched on from the hardware, hence the +48V label above the LEDs. Phantom can be switched on/off individually for any channel, and to do this on the hardware you use the main encoder to select phantom (the default option on startup) and scroll to the desired bank »

CB Electronics XP-Relay
$2200

PROS

- The cleanest signal path possible!
- Supplies and passes through +48V phantom power.
- Sophisticated software control, with USB or Ethernet connectivity, and native Elgato Stream Deck support.
- Hardware can work standalone.
- MIDI control opens up lots of possibilities for artists and engineers.

CONS

- None really — though sync with a DAW plug-in would be the cherry on the cake!

SUMMARY

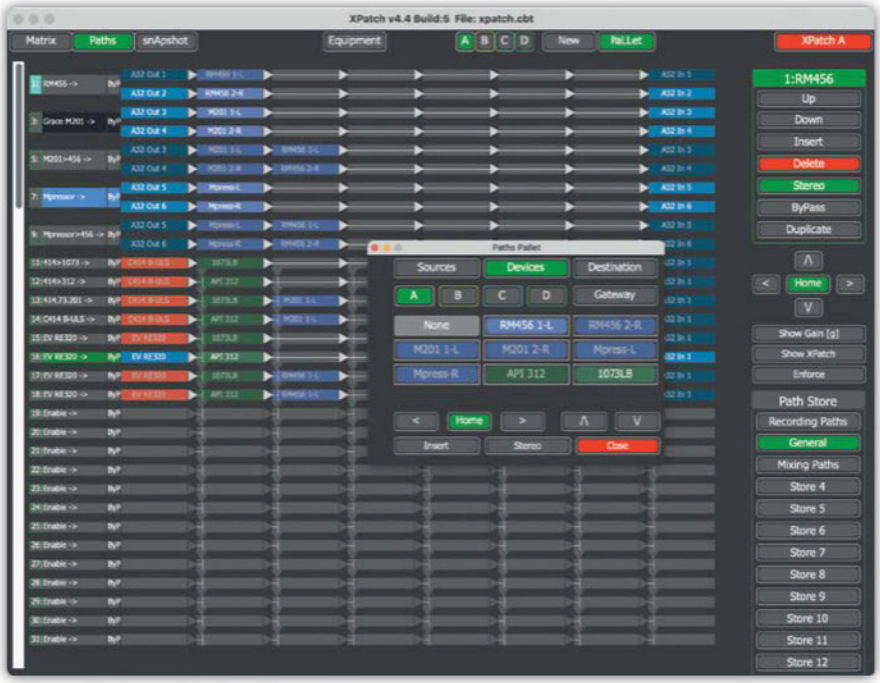
An excellent digitally controlled analogue routing matrix, which, with nothing in the signal path other than relay switches, can pass through (or provide) phantom power. Once configured, it can be controlled from its own Mac/Windows app, or remotely using MIDI.

» of inputs (1-8, 9-16 or the two front-panel XLRs), then hold down the relevant channel button while pressing the encoder. You can also operate phantom from the control app. Phantom power provision is welcome and convenient, especially if, as I do, you have some nice vintage preamps that don't supply it, or have multi-channel devices where it's switchable only in banks. The presence of the encoder, though, means there's scope to give these buttons secondary (or multiple) functions. During my review tests, I was in frequent communication with Colin Broad (the CB behind the company name) and suggested that they might be used to select routing presets — apparently I wasn't the first person to mention this (great minds...) and within a day or two he'd already implemented it! How's that for responsiveness?

Softly Does It

The XPatch app is standalone software, by which I mean there's no dedicated DAW plug-in version. At least not yet; I gather CB are looking to secure the services of a plug-in developer. In the meantime, if recall from a DAW project is particularly important to you, CB were, at the time of writing, about to release an interim solution using MIDI SysEx messages. The XPatch app is accompanied by another utility called MIDI UPD, which is used for setting up the MIDI channels for any attached XP-Relay and/or XPatch devices, and for installing firmware updates.

The general approach and layout of the XPatch software is broadly the same as when I reviewed the XPatch 32, but it has evolved considerably. There are four main pages and to start, having hooked your gear up to the XP-Relay, switched on the latter and initiated the software, you go to the Equipment tab and define what hardware is attached to what I/O channels. You can specify whether the equipment is to be treated as a Source (something connected to an XP-Relay input only, not an output — a mic, instrument or audio interface output, for example), a Destination (something connected to an output only, for example a compressor's side-chain input or active speakers), or a Device, which you've connected to both an input and an output (eg. a reverb unit or dynamics processor). This classification dictates how the gear is listed in the Paths and Matrix windows discussed below. In the Equipment page, you assign the XP-Relay channel numbers to your equipment, along with a short name that will appear on other pages and the hardware, and, optionally, a few other details. It's all pretty intuitive, though if you have only the XP-Relay, you'll find, alongside many useful facilities, some superfluous ones that apply only to the XPatch hardware, which could be confusing for new users.



The Paths page allows you to create preset chains of gear, with those sharing the same Destination deactivated as another is activated.

With your equipment suitably specified, you can perform basic routing very easily from the Matrix page, which is laid out as a grid with Sources listed on the left and Destinations along the top. You simply click a square to link any source to any destination. A reassuring relay click, along with the a green dot in the square you clicked, confirm that the desired connection has been made. Another little detail added after speaking with Colin is that, as of v4.4, you can right-click on a square in the Matrix page's phantom power column to activate/deactivate phantom power for that Source.

The real power of this software, though, lies in the Paths page. Again, this is laid out in a grid-like fashion, with lines and triangles to indicate the signal routing. You can create up to 128 routing presets called Paths in each of the many banks (an arrangement that will be familiar to anyone who's worked with MIDI instruments and effects before), and click on them to (de)activate them. It's possible to feed multiple paths from a single Source, and as long as two Paths don't employ the same Devices or Destinations they can be active simultaneously. Activating a Path that shares a Destination with an already active Path deactivates the earlier routing. Also, note that you can use a Device as a Source, the idea being that you might want to use some gear, such as a mic preamp, as an input Source or as an insert Device in different scenarios. Creating Paths is much easier than last time I used this software, when I found I needed to rely more on my computer keyboard. Now, you can just click on either the Pallet button at the top or any empty insert 'block', and the Paths Pallet floating window appears. From here, you can drag any Source, Device or Destination to the desired position on any Path.

Each Path connects the equipment in series, with the signal flow running left to right. For mono Paths, there's a dedicated downward triangle at each stage — click this and it mults the signal to the row below, making the creation of parallel processing chains really easy. At present, it seems that you have to (de)activate each parallel Path separately, but I gather CB are considering a grouping facility that would allow the whole parallel chain to be (de)activated with a single click/command. Stereo-linked Paths don't have the mult button (which I suppose could look confusing!), but similar routing is »

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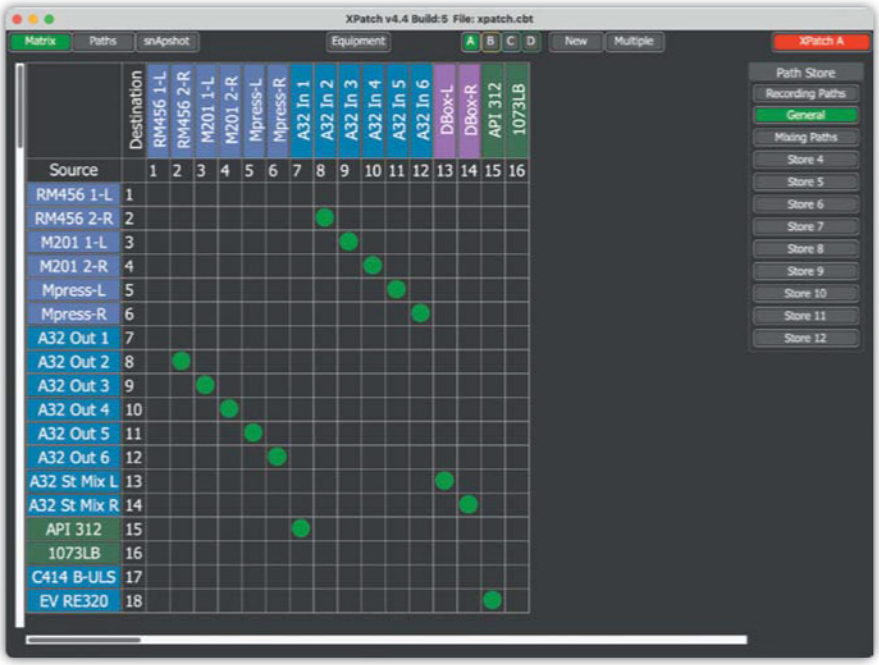

The Matrix page of the XPatch software, which allows you to directly route from any input to any output — each source can feed multiple destinations.

» easy to achieve by creating and activating multiple Paths fed from the same Source. By the way, any routing selections you make in the Paths view will be reflected in the Matrix view.

Remote Control Options

The fourth of the app windows I mentioned is called Snapshots, and allows you to save/load complete routing and phantom power profiles from/to the hardware. It's a less versatile system than the Paths and if you plan on operating it using the software something you might not bother with, but importantly you can load multiple Snapshots to the hardware and switch between them using the footswitch, MIDI or, since a recent firmware update, the front-panel buttons. If the XP-Relay is connected to a router, you can also control it from a phone or tablet, using the TouchOSC app for iOS or Android. So anyone who relies on a few 'standard' routing configurations can load them onto the hardware, and then do all their switching without ever needing to attach a computer or open the app — and that obviously opens up many possibilities for performers.

Another convenient control option is to use an Elgato Stream Deck (see box) to switch different Paths on/off. It needs to be attached to a computer running the XPatch software to work, but means you never need to leave your DAW to look at the app, which is refreshing.

That the device can be configured using MIDI also opens up lots of options. For example, one user has set up an inventive MIDI-controlled XP-Relay system in which he places Bluetooth Flic Buttons (which can transmit MIDI) alongside his various instruments, and he can simply sit at an instrument, hit the Flic Button to enable the 'normal' routing for the instrument, and start playing. It seems more MIDI command options are being added too — for example, CB Electronics recently implemented commands to Disconnect Destination 1, Connect Destination 1 to Source 1, Connect Destination 2 to Source 1… and so on. For anyone with ambitions to get into the nerdy details, this system is brimming with potential for custom control setups.

Verdict

I was impressed with the XPatch but, if anything, I'm even more impressed with the XP-Relay — it's a winning combination of sophisticated software and refreshingly simple hardware. While the passive signal path means it lacks some XPatch functionality, it also opens up really useful options, such as comparing different mic and preamp combinations, without any of the compromises involved in doing that with the XPatches or, say, Flock Audio's Patch. The XPatch app is much improved too, and the remote control options are evolving really nicely, not least the Stream Deck support — operating it from that has been a joy. I'll be interested to see what comes of CB's experiments with DAW integration too. There aren't really enough inputs and outputs on this unit to accommodate my own mixing hardware setup, though two 16x16s could come close and there will be plenty of I/O for many — and there's the option to use this device alongside an XPatch (or three!). Used on its own, the XP-Relay would a great option for a lot of engineers, musicians and performers, whether on stage or in the studio, and it could be superb in an analogue or hybrid mastering setup. The price isn't bad considering what's on offer, either — so if an automated analogue patchbay holds appeal, the XP-Relay has to be worth auditioning!

Elgato Stream Deck Support

Stream Decks, macro keyboards whose keys house tiny screens to display icons or text, are usually configured with Elgato's own software, but in order to show the names used in their XPatch app CB programmed their app to support them natively. A standard (5x3) or XL (8x4) Stream Deck can be used to select Paths (signal chains defined in the Path window), and you can change banks too. There's no icon support yet and the colour scheme is monochrome, but CB are exploring options to develop this side of things. The main benefit is that you don't have to flip between your DAW and the XPatch app to enact your routing changes; the only downside is that you must close the Elgato software while using a Stream Deck with XPatch.

The XPatch software now includes built-in support for Elgato Stream Deck keyboards, which can be used to activate and deactivate any of the routing Paths that you've created in the app.

$ 16x16 model, as reviewed: $2200.
T Wes Dooley Audio +1 626 484 4887.
E stereoms@aol.com
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Sound Radix Auto-Align 2

Automatic Time & Phase Alignment Plug-in

This powerful alignment tool is now easier to use and boasts new functions — along with impressive results.

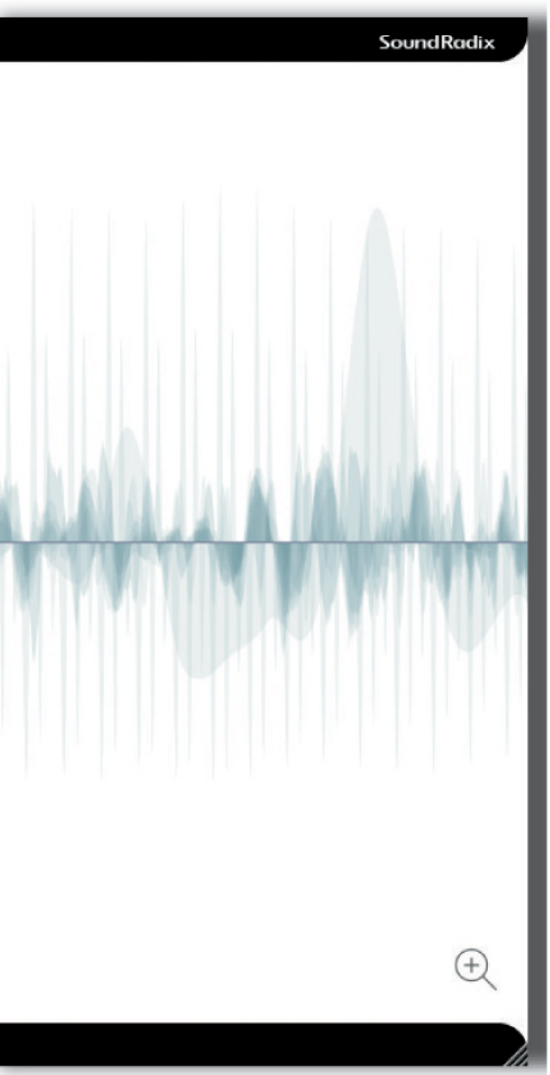
NEIL ROGERS

Sound Radix tend to create original and innovative plug-ins that address common problems in audio production. I've reviewed some of their software in *SOS*, and often use their excellent Surfer EQ and Drum Leveler plug-ins in my mixes. Under review here is an updated version of probably their best-known, most widely used plug-in: Auto-Align. A time-alignment tool, it's intended to help engineers optimise the phase relationship between multiple mics/DIs used to capture the same source. Auto-Align 2 runs on Intel/Silicon Macs (macOS 10.9 and up) and Windows (10 and up), and supports the usual plug-in formats as well as ARA2. For this new version, Sound Radix have rewritten their algorithms, added important new features, and made it far quicker and easier to use.

Mic Interactions

Before I assess Auto-Align 2 itself, it's worth spelling out the issues it aims to address. When we record anything with multiple mics, each one will pick up the same sound (broadly speaking) at slightly different times, due to the different distances at which they're placed relative to the source. When the mics' signals are mixed together, these timing differences mean that the similar signals aren't fully in phase: there will be some degree of comb filtering, often perceived as a 'hollowing out' of a sound or a lack of 'fullness', and transients can sound a bit soft or smeared.

This can apply to anything where you use more than one mic simultaneously, even if using separate mics on different instruments in close proximity. For example, with a singing guitarist you might have one mic on the guitar and another



Auto-Align 2 automatically groups related tracks on which an instance of the plug-in has been inserted, and aligns the time and phase of all tracks internally within the group.

on the vocal; each mic will pick up some of both sounds. But the classic example is a multi-miked drums: a kit will typically be captured with a selection of close, overhead and room mics, and there will be timing differences between the various mics, with the close mics picking up spill from the other kit elements and each hit inherently being earlier in time than in the main overhead/room mics. The more mics you use, the more complex the time and phase relationships become.

As you gain experience as a recording engineer, especially if you have the luxury of a settled studio setup and good monitoring, you'll develop an ear for how the mic signals interact, and learn to adjust mic placement accordingly, carefully moving them around and listening for changes. You'll also learn to check the effect of inverting the signal polarity, and you might even start measuring the distance between the key mics and kit pieces (particularly the snare and overheads) to minimise problems between them. But, eventually, you'll reach the point of diminishing returns (and impatient artists!) and must hit the Record button. And, of course, we're not always at the recording session; increasingly, I'm asked to work with recordings made in a home studio, where the limitations of the space, monitoring and perhaps gear usually mean there's limited scope to fine-tune the mic placement.

Either way, once the signals have been committed to ones and zeroes in our computers, we might still hear problems and feel the need to refine the relationship between the tracks by nudging the audio for different mics forward or back, and experimenting with polarity inversion. This can sometimes deliver improvements but it can be very time-consuming and often unsatisfying: given the number of individual mic-to-mic relationships, an improvement in the sound of one kit piece can easily have unintended consequences for another.

Instant Fix?

Wouldn't it be great if there were a quicker, easier and more effective way for people to make all these tracks work together better? Auto-Align aims to be just that. The original version was a real-time plug-in (AAX, VST, AU) that required you to place a separate instance on every track you sought to align. You specified which instances were references, and which were to bring the audio into alignment with those references, choosing to align in terms of time and/or signal polarity. The timing was sample accurate, and while the routing and detection was only possible in the real-time version, Pro Tools users could also copy the results to an offline AAX AudioSuite plug-in for processing. That it quickly became popular is not a surprise — it was a much more efficient way of performing a task that had hitherto been a painstaking manual one — but there were some mental gymnastics involved in organising the multiple instances to secure the very best results and, as with manual nudges and polarity inversions, there were still some issues that it struggled to address.

In Auto-Align 2, Sound Radix have simplified the user experience considerably, as well as providing greater functionality and helpful visual feedback. Notably, they've added ARA2 to the list of plug-in formats, which is welcome news, since Auto-Align 2 is intended for use at the beginning of the signal chain. The 'listening' process is now only required when using the non-ARA version of the plug-in — the real-time version of the plug-ins must be allowed to 'listen' to at least 30 seconds of audio. But capturing can also be done faster than real time by simply bouncing the project's range: when playback is stopped, either by the user or when the bounce is complete, Auto-Align 2 will automatically analyse the audio and apply its processing.

Another key new feature is the inclusion of all-pass filtering, or 'phase rotation' as it's sometimes known. This allows the algorithms to optimise the phase relationship between tracks with far greater precision than can be achieved with a combination of nudging and polarity inversion alone. There's also revamped visual feedback, with options to display what AA2 is doing as waveforms or a correlation meter.

In their promotional material for the release of AA2, Sound Radix recommend putting an instance of Auto-Align 2 as the first plug-in on every track in your session. There's nothing really to be gained from putting it on any tracks that have no time or phase relationships with others, but neither will it do any harm to »

Sound Radix Auto-Align 2
$199

PROS
- Very quick and easy to use.
- The automatic grouping function works well.
- Time and phase alignment processing is very impressive.
- Good, clear metering.

CONS
- None... though arguably it could encourage lazy recording decisions!

SUMMARY
Auto-Align 2 is a significant update of an already handy tool. The workflow has improved considerably, and as well as optimising a good recording, this new version can drastically improve the quality of a less-than-stellar one, and even opens up more creative use of multi-miking in the studio.

» tracks it doesn't need to process, and putting it on every track would avoid you having to think about which tracks may benefit and which won't.

With your multiple instances in place, there's none of the faffing around the original version required. You don't need to decide and specify which tracks need to reference which others: you simply hit the Align button in any one instance. Auto-Align then sorts the various instances into 'correlation groups' based on what its analysis suggests they have in common, and attempts to correlate all the signals in the group with each other, aligning everything to the earliest signal in that group, and then applying polarity inversion/phase shift as required.

For a traditional rock band mix, for example, Auto-Align 2 would typically place all the drums into one group, and align the other tracks to the timing of the kick or snare close mic. It would create another group for the mic and DI of a bass, and align to the DI. It would make another for any guitar part recorded with multiple mics… and so on. There are no interactions between the different groups, but the timings we're dealing with here aren't going to interfere on a musical/rhythmic level, and it's also possible to go in and reassign tracks to different groups manually should you feel the need.

With a bare minimum of user input, then, AA2 can automatically optimise the time differences between your mics in a way that ensures maximum reinforcement and minimum cancellation. It saves a lot of time compared with manually nudging tracks into place, and if that's all you need to do it could quickly earn its keep. But the real impact of this tool lies in its ability to manage the complex phase relationships between different mics. Described by Sound Radix as a "spectral phase optimisation" stage, all-pass filters are used to shift problematic frequencies on one track to bring them into alignment with corresponding frequencies on another. This is static filtering, similar to that used in Auto-Align Post 2 to compensate for the use of a fixed high-pass filter being used on one mic but not others. We don't have the ability of that plug-in or Sound Radix's Pi (both of which are worth checking out!) to dynamically manage the phase rotation, but a benefit is that after the initial alignment the plug-in has done all its work, so your CPU is freed up for other things.

The correlation meter view, which takes the place of the waveform view in the main picture.

In Practice

That's the theory, but how good have I found this plug-in in practice? As soon as I got my licence, I downloaded and installed AA2 (a painless process) and put it straight to work on a mix I was doing for a rock/emo band I'd recorded in my studio. I was especially keen to hear what it would do to my drum recordings. I take great care when recording drums at my studio, and wasn't looking for AA2 to 'repair' anything, but I had to admit that it made the close and outside kick drum mics sound fuller and more punchy — definitely a worthwhile improvement. The toms also seemed to benefit a lot from the processing, and I was struck by how much more pronounced the stereo imaging was when the drummer moved from the rack to the floor tom during fills. Overall, there was a clearly audible positive effect on the drum sound as a whole, and the thing I was most pleased about was that it seemed to sound natural, artefact-free and generally not 'mucked around with'. Not bad for the push of a button!

"The results were much more dramatic and impressive than I'd expected…"

The other source in the same mix that had the potential to benefit from AA2's processing was the bass guitar, for which there were separate tracks for the close-miked amp, a clean DI, and a stereo room mic that I'd left up from the drum tracking. I felt that this was a good test for AA2, because before any processing I could hear some interesting 'phase issues' when bringing up the room mics that resulted in a pleasing, full-bodied bass sound. After AA2's processing, the combined bass channels seemed to have less of the 'subby' low-end, but the bass did feel more solid around 75-150 Hz — subjectively, I'd say there wasn't really a clear 'winner', as there had been with the drums, but AA2 did present me with a different and still appealing option which I could decide upon in relation to the rest of the mix.

I also tried AA2 on a few other projects, in which the recordings were a bit more delicate and less drum-heavy. On an acoustic guitar recording, I liked how it naturally seemed to pull together different microphone options from the same take and make it more viable to use a combination of channels and

adjust their balance without unexpected changes in tonality. Again, the differences were most audible towards the lower end of the frequency spectrum, which is where phase problems often tend to present themselves. Getting into even more delicate territory, I used AA2 on a harpsichord recording session. I was experimenting with finding the best way of capturing the sound of an instrument that I wasn't very familiar with, and found that AA2 worked superbly for combining two mics that were focused on different aspects of the instrument.

I also made a point of seeing what AA2 could offer me on some material that someone else had recorded in a home studio, and which I'd been tasked with mixing. The results were much more dramatic and impressive than I'd expected, and with a drum recording on which, frankly, everything was out of phase, it made a huge difference to every aspect of the drum sound. The snare sounded much fuller, the low end of the kick drum became powerful, transients generally were much more pronounced, and the sense of stereo placement was dramatically improved. Very impressive! AA2 is not a reason to throw good engineering (or studios) out of the window, but it *is* a great tool for bringing a sub-par recording up to a workable quality level.

Stars Align?

While I'd been impressed with the original version of Auto-Align, it could sometimes be a faff to set up and I was sometimes left with the sense that I could hear subtle artefacts, where the processing struggled to make the most of the recordings. I had no such sense with AA2, which seems better in every way.

The automatic grouping saves so much time, and the results have so far always been good. For those working with material that hasn't been professionally recorded, this plug-in could have a hugely significant impact, since its ability to improve recordings laid down in less than optimum settings is very impressive indeed. But what most impressed me was the subtle improvements it can make to already good recordings. Without wishing to blow my own trumpet, I'm a drummer as well as an engineer and have learned over many years to hear how phase problems affect mic setups and recordings. For a user like myself, AA2 is a simple, reliable option for subtle but worthwhile 'enhancement' — I've used it on every project since downloading it! It also has plenty of potential as a tool that allows you to get more creative at the recording stage, for example trying mic options you previously couldn't get to work phase-wise, or experimenting with more assertive EQ. Drums may be the most obvious candidate for AA2 treatment but it works great for guitars, acoustic instruments and, in fact, any source recorded with multiple mics or channels. It's a plug-in that is very easy to use and well worth demo'ing on your next project. Just promise me you won't let it lead you to make lazy recording decisions!

$ Full price $199. Upgrade from v.1 $99.
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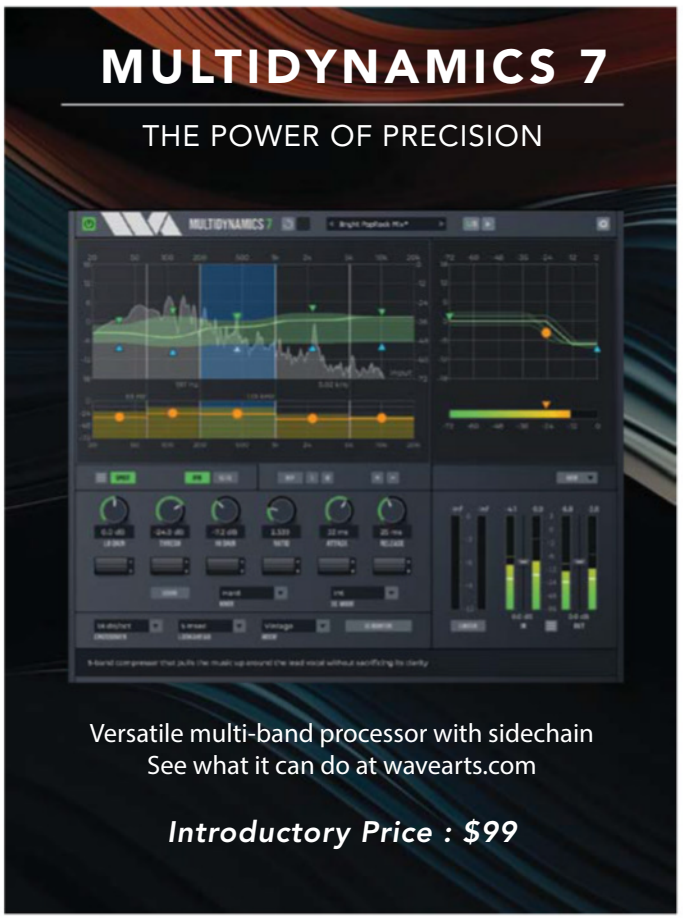

MARK GORDON

Xvive T9

In-ear Monitors

Can off-the-peg IEMs really compete with custom moulds?

There used to be a time when amateur musicians would look in envy at professional bands on big stages at the end of the set, when their in-ear monitors would be pulled out and hung around their neck like a medal — one awarded to only the chosen few who could afford to hear themselves in a live environment. At the lower end of the gigging food chain, we were surrounded by a mishmash of old speaker cabinets and feedback!

These days, in-ear monitors, or IEMs, are no longer the exclusive preserve of the professional musician. They have become far more affordable and, with the advent of cheaper digital mixers and wireless systems, are now far more realistic proposition for semi-pro and amateur musicians. IEMs can improve your performance (you can hear what you play and sing and don't have to fight against a loud band) and save your hearing. If, like me, you are a drummer playing to click or backing tracks, it's almost impossible to work without them.

Founded in 2014, Xvive are relative newcomers to the pro-audio arena and not a name I was familiar with, but they appear to be making inroads with their wireless technology, which includes the U4 In Ear Monitor Wireless System. To complement this system, they've launched their own in-ear monitors, the T9s. Designed and hand-finished in the USA, and featuring dual balanced-armature drivers (read on for more on this design approach), the T9s profess to offer "pro-level audio quality previously only available from custom in-ears, at a much more accessible price". Having used custom-mould IEMs for many years, I was keen to find out if the T9s were a worthy alternative.

Xvive T9
$189

PROS
- Great build quality.
- Balanced sound.
- Impressive isolation.
- Detachable cable.

CONS
- May be a little large for some people.

SUMMARY
If you'd like to try in-ear monitoring but aren't ready to splash out on custom moulds, the T9s are a great option, offering exceptional build quality, great sound and impressive isolation at an affordable price.

Box Your Ears

Arriving in a neat white box, the T9 package comprises a hard-shell zip case, cleaning tool and 3.5mm-to-6.5mm headphone adaptor. The earpieces themselves come mounted in a large foam block for protection, beneath which is stored a 1.5m rubberised cable that attaches to each earpiece via a two-pin connector. A three-inch plastic sleeve surrounds the cable at the point where it attaches to the earpiece, making it curve slightly and enabling it to fit neatly around the top of your ear. Having a detachable cable is always a bonus — no need to get new headphones if the cable fails or is damaged. A selection of both silicone and memory foam earbuds are provided, in three different sizes, with the silicone buds coming in both red and yellow, so you could use a different colour on each side to more easily identify right and left.

Constructed from a clear resin, the earpieces feel very high quality. There are no obvious seams where they might come apart or break, and the cable connects with a reassuring click and feels very securely attached. The transparent design not only looks great but also allows you to see the internal workings, which include two balanced-armature Knowles drivers — one powerful, high-headroom bass/mid balanced armature, and one custom-tuned and custom-ported micro tweeter. Unlike dynamic drivers, which use a coil and magnet to create sound, balanced-armature drivers vibrate a tiny reed balanced between two magnets inside a tiny enclosure. The motion of the reed is transferred to a stiff diaphragm and on to an output port, which can be tuned for a specific frequency range. Its compact design makes it ideal for in ear-monitors, and allows the use of multiple drivers to handle different frequency ranges, achieving an overall flatter response than you might get with dynamic drivers.

Fit & You Know It

When I first took the earpieces from their foam packaging, I was immediately struck by how large they are. Even with them inserted deep into my ear canal ,they protrude quite a way. Whilst this didn't cause me an issue, it may prove to be problematic for someone with smaller ears.

After experimenting with the earbuds, I opted for the smaller memory foam buds. It is well worth spending time to ensure you get the best fit possible, not only to ensure a secure and comfortable fit but also because poorly fitting IEMs will compromise

the performance of the system, especially in the lower frequencies.

The next step was to listen to some audio. Be warned, these IEMs are LOUD! If you're switching to the T9s from another set of headphones, I strongly suggest you turn the source volume down and work your way up to a comfortable listening level. Several factors have a bearing on how loud headphones are, but the sensitivity is a good basic guide. The T9 spec quotes 120dB at 1kHz, whereas my ACS custom moulds are rated at 112dB and the Shure SE215s at 107dB. You don't need to drive these IEMs hard to get a good listening level.

Playing back a selection of pre-recorded material from a headphone amp, I thought the T9s performed very well, giving a smooth and balanced sound. The bass was full and the high frequencies cut through very nicely. There's no harshness to the sound and, if anything, that gives the T9s quite a hi-fi quality.

The second test was to use the T9's live. I play in a function band where I regularly use in-ear monitors (the ACS Evoke 2 custom moulds I've already mentioned), so it was simply a case of swapping headphones. Having learned from my earlier experience, I reduced the volume of my monitor setup considerably to compensate for the T9s' sensitivity.

The sound isolation provided by the T9s was excellent, certainly on a par with my custom moulds. If you're new to IEMs, the feeling of detachment from the outside world can take a little getting used to, but it does ensure the in-ear mix you create isn't compromised by external noise, particularly on a loud stage. This is especially important when playing drums and playing to a click.

I noticed more low end in my mix than I'm used to, but not to the point where I found it an issue — bass remained clear and defined. As with the pre-recorded material, the highs were very much present but not harsh, which makes for a much less fatiguing experience during a two-hour set. I found the T9s to be very comfortable and secure, which is also an important factor if you're wearing them for long physical performances.

I Will Xvive

It's almost impossible to explain exactly how a pair of IEMs sound, as using them is an experience that's unique to the wearer. Several factors can affect your perception, with fit being the primary one. Having used the T9s now on several gigs, I'm very much enjoying them. They are a great fit, and very comfortable; the isolation is on a par with my custom moulds, and the dual balanced-armature drivers give an even sound with an extended bass response and a clear and detailed high end. This makes extended listening particularly pleasant, and the high sensitivity means there will be plenty of level, regardless of what you are using to drive them.

"The isolation is on a par with my custom moulds, and the dual balanced-armature drivers give an even sound with an extended bass response and a clear and detailed high end."

The detachable cable is a plus point, and I've also found that its rubberised quality prevents it from tangling up. The earpieces themselves look as though they will easily endure the rigours of life on the road — which is, again, an important consideration.

Most IEMs in this price range use either two dynamic drivers or a hybrid combination of a dynamic and balanced armature. The ACS Evolve, with its three Knowles balanced-armature drivers, would cost closer to £650 once you factor in the cost of getting ear impressions for the custom moulds. So I'd highly recommend the Xvive T9s to anyone looking to enter the world of IEMs, as they do, indeed, appear to offer something comparable to custom-mould quality at a far more affordable price. ■■■

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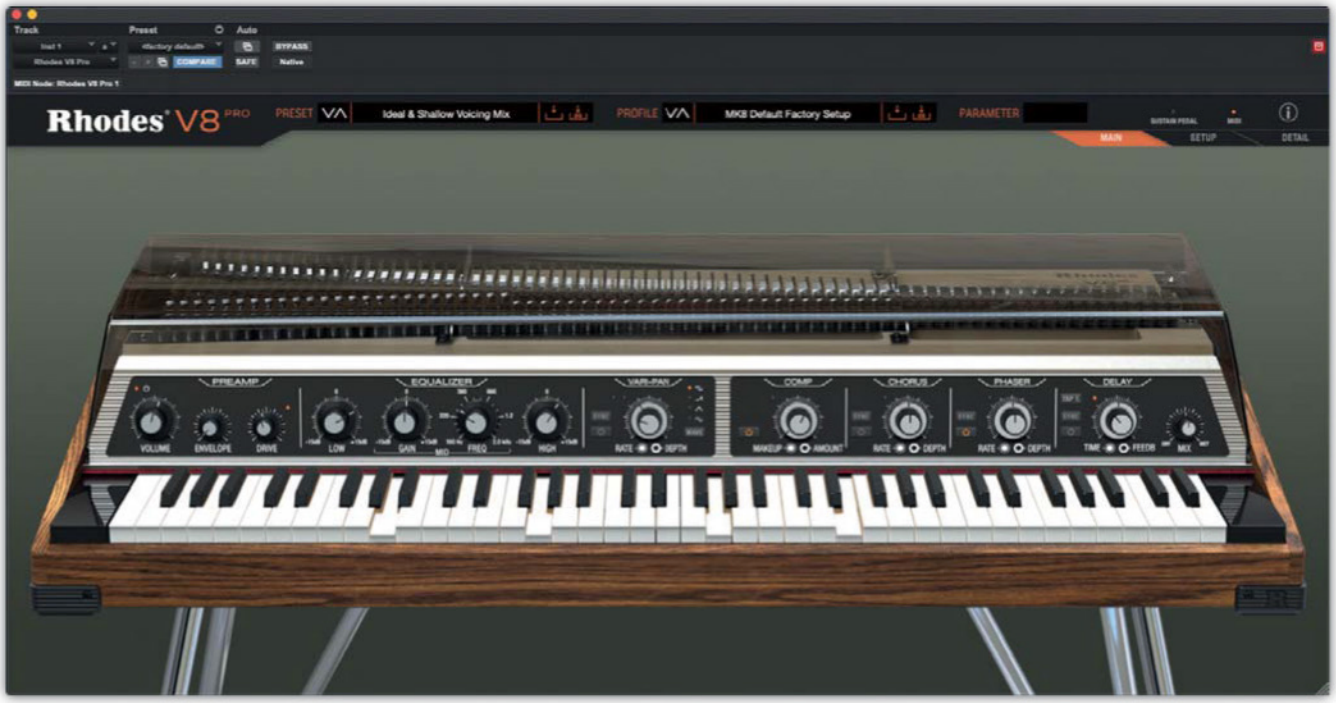


Rhodes V8 & V8 Pro

Software Instruments

Rhodes' software recreation of their flagship instrument shows all the same painstaking attention to detail.

SAM INGLIS

From the Minimoog to the Mellotron, classic keyboard instruments are being reissued and reinvented for the 21st Century. The most impressive reintroduction so far has to be that of the Rhodes piano. A labour of love driven by the passion of Rhodes tech and keyboard ace Dan Goldman, the Rhodes Mk8 is a thing of beauty. That's partly because it's not a slavish recreation of any previous Rhodes model. Dan has drawn on his unrivalled experience and knowledge to fine-tune the design of almost every component, from the tines to the dampers, and hasn't been afraid to improve on weak spots such as the clunky sustain pedal mechanism in the original models. The Rhodes Mk8 is also a rare example of British manufacturing at its finest, with everything apart from the keybed being made in the UK.

The inevitable consequence of Dan's uncompromising approach, though, is that the Rhodes Mk8 is not a cheap instrument. The price tag hasn't stopped it being wildly popular (and back ordered), but does place it out of the reach of many. So, for those of us who would love to own one but can't afford or justify the cost, Rhodes have now made available an official software version.

You Take The High Rhodes

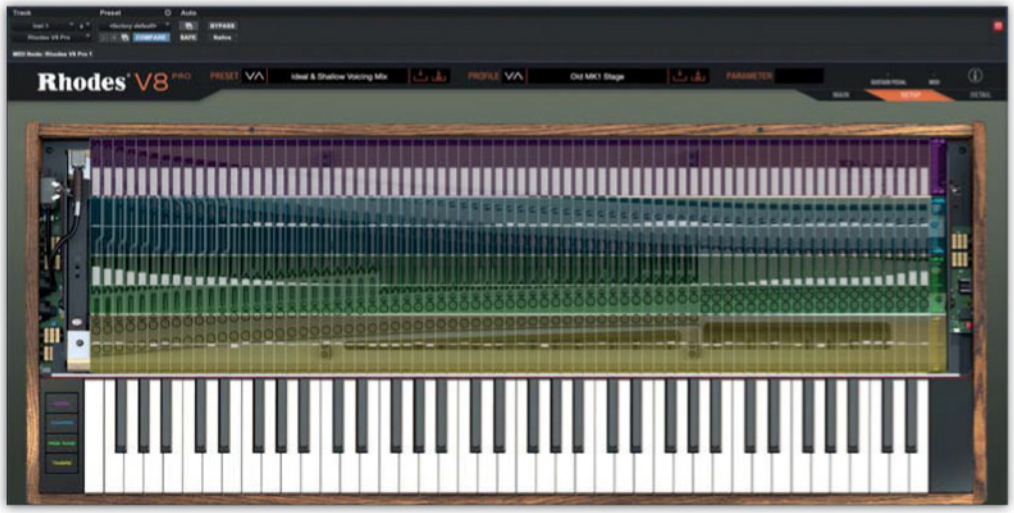
The Rhodes V8 plug-in comes in two editions. These, to some extent, mirror the two different versions of the hardware instrument, which can be ordered with or without onboard effects. At the time of the *SOS* review in January 2022, the FX version was not quite complete, though the potential of the compressor, phaser, chorus and delay was already obvious from demos.

In plug-in land, the version including effects is known as Rhodes V8 Pro. Although there isn't (yet?) a 'suitcase' version of the Mk8, the Pro plug-in also includes amp, cabinet and microphone modelling. Other aspects of the Rhodes Mk8 electronics have also been simplified for the basic version of the plug-in, and features such as the envelope follower and auto-pan waveform selection are only available in V8 Pro. Finally, V8 Pro allows you to dive under the hood and adjust the tuning, timbre, level and damping of individual tines in a way that's not possible in the basic plug-in.

The Rhodes V8 virtual instrument is available for macOS and Windows, and supports all the major plug-in formats, though there is currently no standalone version. Initially, you install a small downloader app, which in turn pulls about 13GB of sample data from a server (for the Pro version). This process took quite a long time for me, so I suspect there are some bandwidth limitations at their end. Once installed, it's activated by entering a licence code.

Three Way Split

One of the things that makes the hardware Rhodes Mk8 so desirable is its industrial design. Overseen by Axel Hartmann, it's a triumphant hybrid of classic and modern, and is also extensively customisable. Rhodes are naturally proud of this and so have attempted to replicate the same look in software, using a near photo-realistic representation of the keyboard and front



Only available in the Pro version of the V8 plug-in, the Setup page allows per-note tweaking of parameters such as fine-tuning, Timbre Shift and damper response.

panel, with the internals visible through a virtual smoked plastic top. The on-screen keys are depressed in response to MIDI input, but hammers and damper arms don't move.

The plug-in GUI can be freely resized by clicking and dragging in the lower right-hand corner, and Rhodes have also sensibly upped the scale of the control panels. However, there's still a lot of unused space, and without expanding the window to its largest supported size, I struggled a little to read the legending and reliably target controls with the mouse pointer.

In the V8 Pro plug-in, the GUI can be switched between three views. The Main view presents pretty much what you'd see if you were sat in front of a real Mk8 with the effects board: preamp, EQ and vari-pan settings on the left, and effect settings on the right. Detail view presents a number of additional controls relating to all aspects of the piano. Some are global settings such as overall tuning, velocity and pedal response, and the level of mechanical noise, while others are mostly secondary effects parameters that are not available from the Main view. You also need to visit Detail view if you want to set up amp and cabinet simulation.

A minor irritation with this split layout is that there's no one place where you can see all the effects parameters at once. The Detail page shows only the additional parameters that are not visible in the Main page, so if, for example, you want to adjust the Resonance of the phaser, you need to go to the Detail page — but you can't actually turn the phaser on and off there, or set its rate.

The third, Pro-only view is labelled Setup, and hands you a virtual screwdriver to adjust the behaviour of individual notes. Level, damper response, fine-tuning and Timbre Shift can be set on a per-note basis for each of the 73 notes by clicking in the appropriate box and dragging the mouse up or down. Keep the mouse held down and drag left or right and you can draw in sweeping patterns across multiple notes. Timbre Shift is an important factor in the overall sound and, according to the manual, changing the setting "controls the angle of the tine relative to the pickup and the distance of the tine relative to the pickup pole, increasing or reducing its harmonic content and perceived level. At higher levels of Timbre Shift, the tine position is lower relative to the pickup pole's centre, bringing out more harmonics and animating the tone more overall. At lower levels, the tine is raised higher above the pickup pole centre and fewer harmonics are present in the resulting, 'rounder' sound."

You can't simply hold down a modifier key and adjust one of these parameters by the same amount for all notes simultaneously, but several of them have their own global controls in the Detail page. Obviously, if you wanted to make the entire piano a quarter-tone flat for some reason, you'd be better off doing so using the global tuning control than by laboriously adjusting each individual note's tuning setting. Less obviously, the global Tuning, Timbre Shift and Level settings are applied on top of whatever you're doing in the Setup window, so you can for example achieve an extreme Timbre Shift by placing the global control at one end of its travel and doing the same for all notes individually. Likewise, the additional EQ that appears in the Pro plug-in's Detail page is additive with the EQ in the Main page.

Profile Pictures

The occasional duplication or crossing of parameters makes sense in the context of the V8 Pro plug-in's dual-level preset architecture. You can store and recall Presets, which encompass all parameters, as you'd expect; but you can also store and recall Profiles, which cover only settings made in the Setup page and the upper row of the Detail page. The idea is that a Profile stores a basic Rhodes configuration, while Presets additionally store what might be considered performance controls, such as effects settings. Thus, for example, the Profile EQ in the Details page is considered part of the underlying sound, whilst the Main page EQ is intended to modify that sound on the fly for effect.

A very healthy selection of factory Presets and Profiles is included, and goes way beyond minor variations on the core Mk8 sound. There are Profiles that aim to match the sound of older Rhodes models, both in good and not-so-good condition. There are Profiles that store stretched, compressed and 'smile curve' tunings across the keyboard. There are »

Rhodes V8 & V8 Pro
From $179

PROS
- Provides a smoothly multisampled Rhodes Mk8 sound, but can also do very decent recreations of earlier models.
- Setup page in the Pro plug-in offers endless configuration possibilities.
- Gorgeous effects.

CONS
- Amp/cab modelling is a bit perfunctory.
- Controls and their labels can be hard to make out unless the GUI is at its largest size, and effects parameters are split over two pages.

SUMMARY
The Rhodes V8 plug-in won't stop anyone lusting after the real thing, but it's a fine virtual instrument in its own right, and arguably even more versatile from a sonic perspective.

The Detail view provides access to the V8 plug-in's amp and cab modelling, alongside various global settings and additional effects parameters.

» experimental setups of the kind you probably wouldn't spend days recreating on a real Rhodes, such as Profiles where alternate notes have opposite tuning or timbre settings. And there's a large number of artist Presets and Profiles.

Setting up a real Rhodes piano isn't for the faint-hearted, and I'm sure I'm not the only one who's happy to leave it to the experts, so in a sense, the Setup pane and the factory Profiles make the plug-in much more versatile than the real thing from a practical point of view. In fact, I'd say that if you were contemplating buying a Mk8 piano, it'd be worth getting the plug-in first to get an idea of what sort of tonal range it's capable of, so that you can ask for it to be configured to your tastes. Likewise, if you already own an older Rhodes and you want to match its sound in software, the V8 Pro plug-in gives you most of the tools you need to do so.

One slight oddity about the factory Presets is that they all have the compressor enabled, even those that are intended to emulate older Rhodes pianos that never had compressors built-in. I think the reason for this is that with so many parameters under the hood that can affect the signal level, Rhodes have rightly been cautious about overloads. Disable the compressor, and the output is usually very quiet, to the point where even heavy block chords barely hit -25dBFS on the DAW channel meters. That can be annoying when you need to overdub to an existing backing track, but it's easy enough to correct once you've dialled in the sound to your taste, and preferable to the many instrument plug-ins that hover on the edge of clipping all the time.

Rhodes Trips

Creating a multisampled instrument is a very different project from building an electromechanical piano, but Rhodes Music are well placed to do it. One of their main investors is Matt Pelling, whose previous company was Loopmasters. With access to his vast experience overseeing the creation of sample libraries, you'd expect Rhodes to be able to get this one right, and they have. The core Mk8 sound coming from the plug-in is almost as gorgeous as the real thing. It's rich, plummy, sonorous and balanced — if you want it to be. If you'd rather it was aggressive and biting, it can do that too. Or if you want a more muted, bell-like tone with authentically intrusive thumps and bumps from the mechanism, you can have that at the click of a mouse.

There are existing Rhodes plug-ins that cover an equally wide tonal range, especially those that use physical modelling rather than samples, but what you won't find elsewhere is a painstaking replica of the Mk8's electronics. Rhodes Music have put a lot of thought into the design of their preamp, EQ and effects, and they are quite different both from generic stompbox or studio processors, and from anything that was available as an add-on for older Rhodes models. The Drive, for example, transitions smoothly from gentle added warmth to moderate overdrive without ever becoming gritty or harsh, or losing note definition. Used on its own, the envelope-driven auto-wah is a surprisingly subtle effect, only getting a little squelchy when you turn the EQ's Mid gain right up, but it comes into its own with the Drive and the compressor. Vari-pan is pure bliss at low rates, shading to ring-mod madness at the other end of the scale; again, this is often best used in conjunction with the Drive and auto-wah.

The chorus effect is capable of producing a surprising variety of tones, from rich and floaty to slightly seasick and wobbly, although the need to tab repeatedly between the Main and Detail pages to set it up can get a bit annoying. The same goes for the phaser, which covers much of the same ground in a mysteriously different way and sounds absolutely superb.

Finally, the amp/cab modelling gives you a choice of four amp-plus-cabinet combos, including one based on the original 'suitcase' Rhodes amp. You can also choose whether this should be miked with a virtual Neumann M49 or Shure SM57. With no control over amp settings or virtual mic placement, this feels like a bit of an afterthought, and personally I'd use a third-party amp modelling plug-in in preference.

In all other respects, though, the attention to detail that makes the Rhodes Mk8 such a remarkable instrument in hardware is apparent here too. If you want a great-sounding Rhodes plug-in that you can simply call up and use, the more affordable basic version will do the job nicely. But for my money, the extra features in the Pro edition are well worth the additional cost. You can lose yourself for hours in the virtual setup process without any of the swearing and hard work that's involved in setting up a real Rhodes, and the Pro effects integrate with the basic sound in a way that would be hard to replicate using third-party plug-ins. I still want a real Rhodes Mk8, but as second bests go, the V8 Pro plug-in is a pretty good alternative.

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APS Aether

Active Monitors

There's much more to APS's new studio speakers than meets the eye.

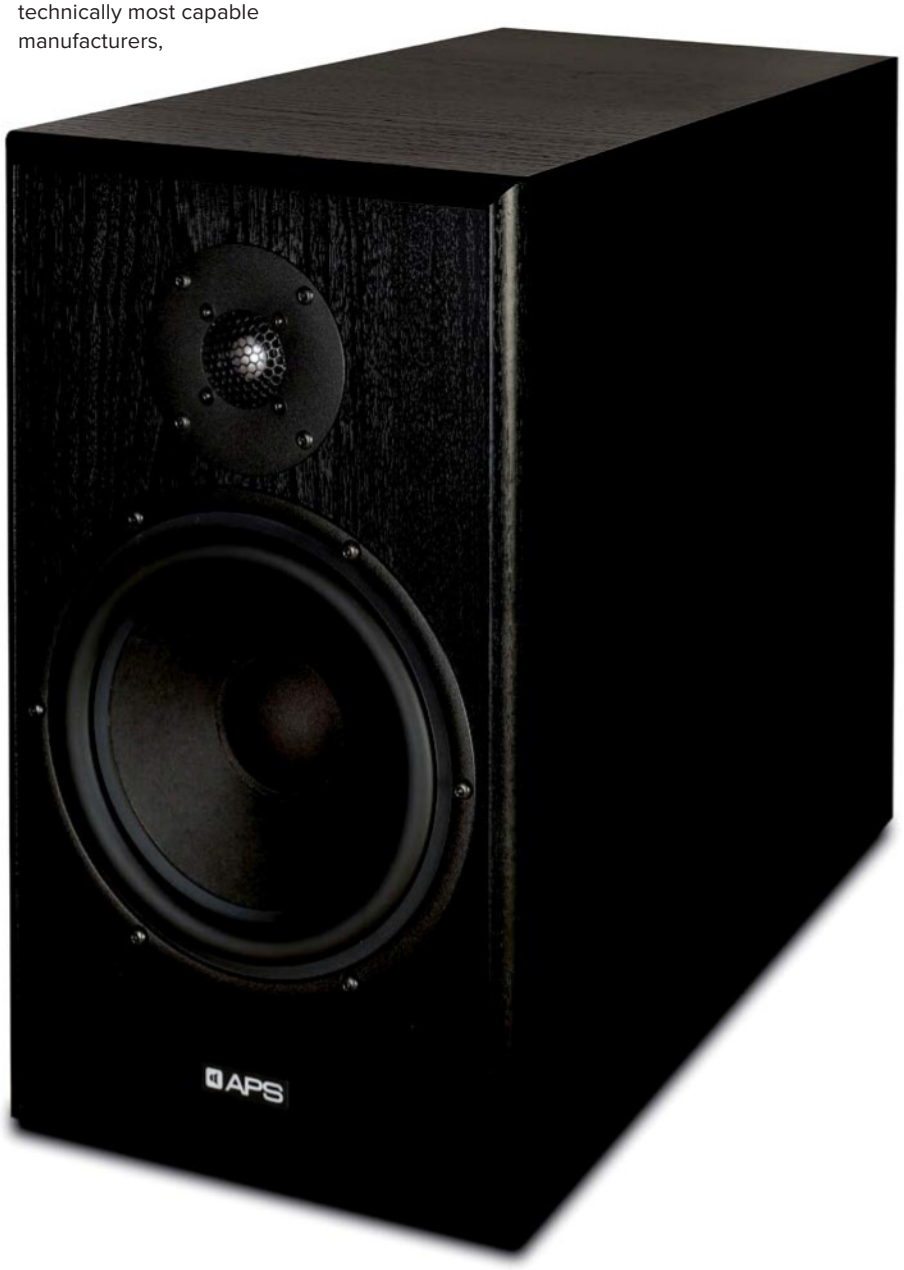
PHIL WARD

I've written about a couple of active monitors from Polish company APS over the past few years, and have been generally impressed with their performance. The entry-level Klasik 2020 model I reviewed in December 2020 was especially noteworthy and, as I wrote at the time, "quite a find!". APS have recently updated their active monitor range, and at the top of said range sits the subject of this review, the Aether.

While the Aether appears in photographs to be of conventional nearfield monitor proportions, APS describe it as a "medium-field" monitor, and now that I have a pair sitting in my studio, I can see why: the Aether is significantly larger and heavier than the typical nearfield product. At 20kg and measuring 44 x 25 x 47cm, it really is quite a significant chunk of monitoring.

In terms of appearance and industrial design, the Aether, sadly, isn't going to win any prizes. Fundamentally, it's a black veneer-finished and internally braced MDF box. Of course, the Aether is first and foremost a professional tool designed to do a job rather than to look attractive (much the same has been said about me), but it's also the case that plain rectilinear boxes are relatively easy to manufacture without a huge investment in tooling. The kind of extravagantly curved die-cast enclosures that, for example, Genelec often employ, are made possible thanks to commercial scale. Smaller manufacturers simply can't stretch to the kind of tooling budget that the likes of Genelec are able to support.

The Aether is a two-way active monitor with drivers that comprise a 220mm-diameter bass/mid unit and a 19mm dome tweeter. The tweeter is an OEM device sourced from one of Europe's longest established and technically most capable manufacturers, and features an aluminium/magnesium dome with a notably generous soft polymer surround that helps ensure that the dome's resonant frequency is kept reasonably low: in this case, more than an octave below the system's 2.8kHz crossover frequency. The generous surround also contributes significantly to the tweeter's radiating area. A quick



APS Aether
$3750

PROS
- Good, uncoloured overall balance.
- Highly informative throughout the entire audio band.
- Great bass.

CONS
- None (if you can get past the plain looks).

SUMMARY
APS tend to fly under the radar of big-name monitoring, but the Aether shows again that successful monitor design and manufacturing is sometimes as much about electro-acoustic skill as it is about big R&D budgets and advanced technologies.

measure of the surround and dome dimensions to illustrate this reveals that, while the radiating area of the tweeter dome alone is around 94mm squared, the surround is only just behind at about 70mm squared. This is not specifically a problem, but it illustrates that when a monitor manufacturer describes a driver dome or cone being manufactured from one material or another, it could be that other moving elements are almost as significant in terms of their acoustic contribution.

Iron Helps Us Play

The Aether tweeter incorporates ferro-fluid within its magnet gap, which serves both to damp its fundamental resonance (the mass of the dome and voice coil bouncing on its suspension) and to increase its thermal power handling (and reduce thermal compression) by providing a thermally conductive path for heat to leave the voice coil and be dissipated in the metallic mass of the the motor system. Ferro-fluid is typically a mixture of an organic solvent with surfactant-coated, nano-scale ferro-magnetic particles — think of it as a magnetic oil. When employed in a tweeter motor system it's injected, usually by syringe, into the gap either side of the voice coil, where it remains, held in place by the flux of the magnet. I've illustrated this in Diagram 1.

The benefits of ferro-fluid sound like a free lunch, but among speaker engineers it has a whiff of Marmite — some love it, some don't. The advantages, as I mentioned above, are damping and thermal control, but the naysayers will argue that ferro-fluid damping introduces non-linearity so is undesirable, and that, if thermal power handling is an issue, it's better to use a larger tweeter. There's sometimes also a concern that ferro-fluid has a finite life; it's said to degrade over a decade or two, leaving the tweeter not performing as intended. But even if degrading ferro-fluid is a genuine concern, a decade or two is a long time for a studio and its monitors, so I'm not sure I'd lose too much sleep over it.

The Aether bass/mid driver is sourced from the same European manufacturer as the tweeter. It incorporates a pressed-paper diaphragm and dust cap paired with a generous rubber roll-surround. APS describe the diaphragm as a 'DKM' item; originally

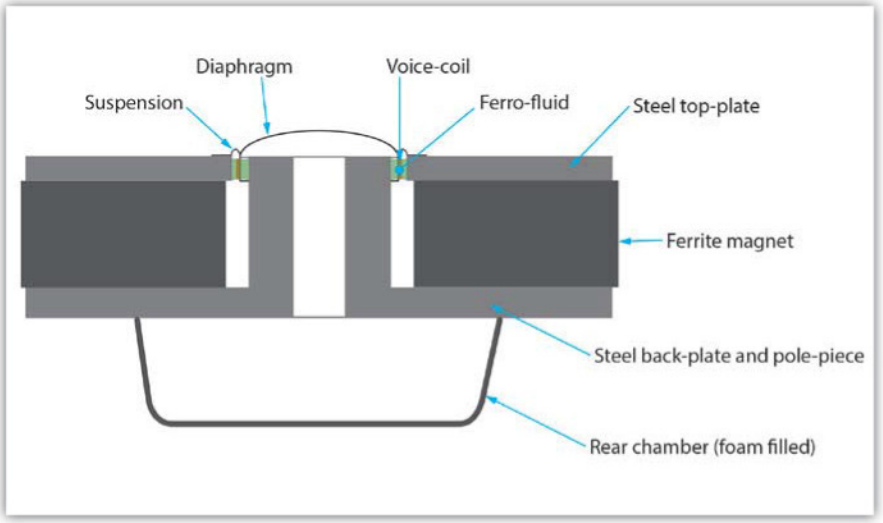


Diagram 1: A cross-section of a tweeter with ferro-fluid cooling and damping.

founded in Berlin in 1938, Dr Kurt Müller, with manufacturing in both Germany and the UK, are almost certainly the world's longest established specialist suppliers of diaphragm and suspension components to the speaker industry. Among European speaker engineers in particular, DKM have become more of an institution than simply a component supplier. If you're designing a driver and not making the diaphragms in-house, DKM will likely be first on your list of potential suppliers, not just for components but also for immense expertise and know-how.

Behind the Aether bass/mid driver diaphragm is a 39mm-diameter voice coil driven by a large ferrite-magnet motor system. The motor system incorporates two copper shorting rings around the pole piece that serve to suppress the distortions that result from magnetic flux and inductance modulation as the voice coil moves in response to the input signal. These kinds of measures taken to reduce driver distortion at source are, I believe, often very significant in terms of a monitor's subjective clarity and ability to play the role of a useful mix tool.

Panel's Labyrinth

Around the back of the Aether enclosure, there's the usual mix of connection sockets and configuration controls, a finned amplifier heatsink, and an extravagantly flared reflex port that APS describe as a "damped bass reflex with APS horn bass reflex technology", of which more in a moment.

The Aether is a fully analogue monitor and its input connections comprise just a balanced XLR/TRS jack combi socket. An XLR output is also provided for downstream daisy-chaining of a subwoofer if required. Aether configuration controls comprise a precision stepped 'volume' control spanning 0dB to -10.5dB, a stepped low-frequency cutoff control offering settings from 30Hz to 120Hz, a tweeter level switch that offers -1.5dB, 0dB and +1.5dB choices, a ground-lift switch, and an unusual approach to room optimisation. Downstream of the input and configuration electronics, the Aether incorporates power amplification rated at 195 Watts for the bass/mid driver and 130 Watts for the tweeter.

The room optimisation functions comprise a frequency control, running from 30Hz to 100Hz in 10Hz divisions, and a stepped level control spanning +3dB to -12dB. The concept behind the optimisation function is that it enables the low-frequency standing wave mode that is the most troublesome at the listening position to be, to some extent, equalised. In the absence of any DSP power within the Aether that might enable digital room optimisation (Sonarworks, for example), I think it's in principle quite a neat idea. I'll report on how well it works a little further down.

Measuring Up

I took one of the APS Aethers to my usual monitor measuring location and fired up FuzzMeasure to see what I could discover. Diagram 2 illustrates the Aether's axial frequency response at one metre, along with its associated second- and third-harmonic distortion. The frequency response is nicely linear »

» — it's neither the flattest nor the lumpiest frequency response I've ever measured, but somewhere towards typical. The distortion performance, at 90dB output level, is pretty good except for an isolated peak around 700Hz — although, even then, the third harmonic level is still less than 1% (there was a time when 1% distortion across the full bandwidth of a speaker was competitive), so the distortion peak in reality is probably relatively benign in subjective terms. The rest of the distortion performance is impressive, the very low level of third harmonic from the tweeter (above 3kHz) especially so.

Diagram 3 illustrates the Aether frequency response variation in the vertical plane. The usual interference dip, where the two drivers go out of phase in their overlap region, is apparent but it is reasonably well controlled and the general similarity of response shape above and below the axial suggests a well sorted crossover design. Also revealed in the vertical off-axis response curve is the expected drop in tweeter level, although this is reasonably gentle thanks to the tweeter's relatively small 19mm diaphragm.

I wrote earlier that I'd investigate and describe the Aether low-frequency loading technique, and I'll start that with Diagram 4. The two curves of Diagram 4 illustrate the result of placing a measuring microphone very close to the bass/mid driver diaphragm (blue curve: ignore the suck-out at 700Hz, it's a measuring artefact) and at the throat of the Aether's reflex horn (red curve). Now, in a conventionally configured and tuned reflex system, the driver close-mic curve would typically show a sharp reduction at the port tuning frequency (usually between, say, 35Hz for large monitors and 75Hz for small monitors). This is the frequency at which the port does most of the heavy lifting in terms of the output of the system. But there is no such reduction revealed on the Aether bass/mid driver curve, just a shallow dip centred around 24Hz. This suggests a heavily damped system with a very low reflex tuning frequency, and this was confirmed to me by APS designer Grzegorz (Greg) Matusiak. Similarly, the red curve of Diagram 4 shows a notably low-Q port resonance that peaks between 20Hz and 30Hz. A close-mic port frequency response curve will typically display a much higher Q. The red curve also shows a complete absence of any resonant features above the port tuning frequency until the output degrades into the noise floor (the noise is the output of the driver — you can't work the port without also working the driver). This is a very good result, because it means little or no undesirable resonant midrange energy escapes through the Aether port.

Tuning a reflex port as low as 24Hz is unusual, and has implications for the Aether generally. A more conventionally tuned port significantly reduces driver diaphragm movement in what is a musically very demanding frequency range, and that doesn't occur here. It means the Aether's bass/mid driver has to work harder. However, the low port tuning frequency, especially in combination with its high level of damping, also means that time-domain errors, in the form of both increased low-frequency latency (technically known as group delay) and resonant overhang, are suppressed. To illustrate this, some further FuzzMeasure analysis revealed that the Aether displays »

The generously flared port exit is mirrored on the inside of the cabinet, and this contributes to its low tuning frequency and unusually good time-domain response.

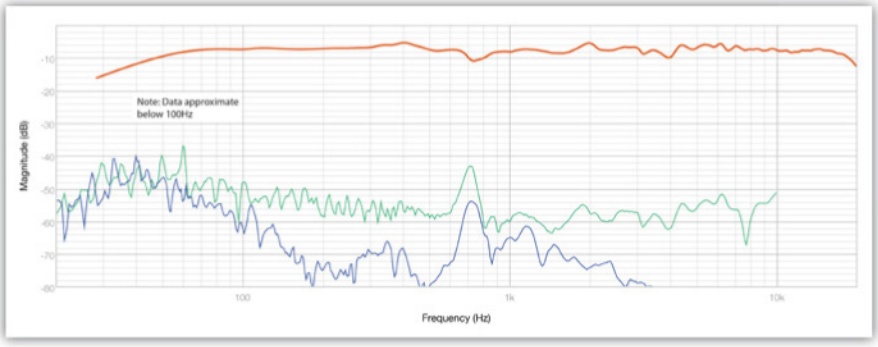


Diagram 2: The Aether's on-axis frequency response, measured at one metre (red trace). The green and blue traces show second- and third-harmonic distortion, respectively.

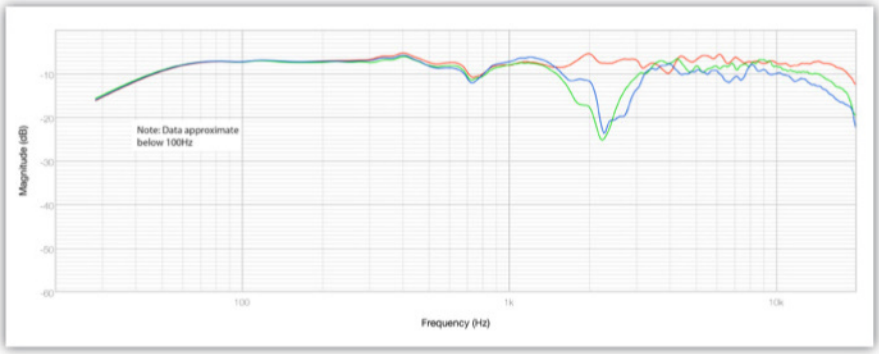


Diagram 3: Comparing the on-axis response (red trace) with that measured 30 degrees above and below (green and blue traces, respectively).

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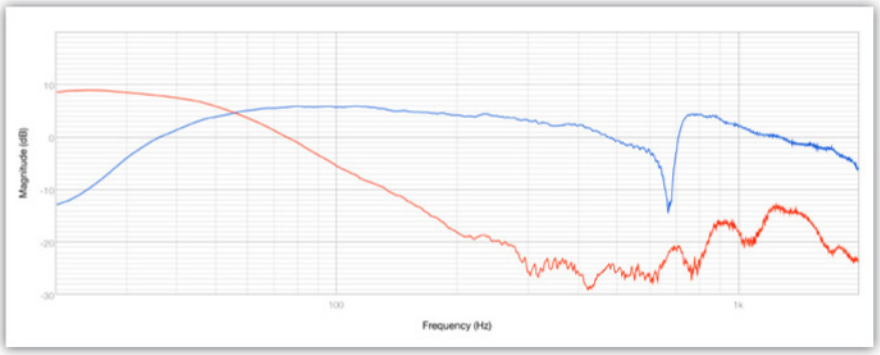


Diagram 4: Close-mic measurements of the bass/mid driver (blue) and the port exit (red), revealing the low Q of the port's resonant frequency.

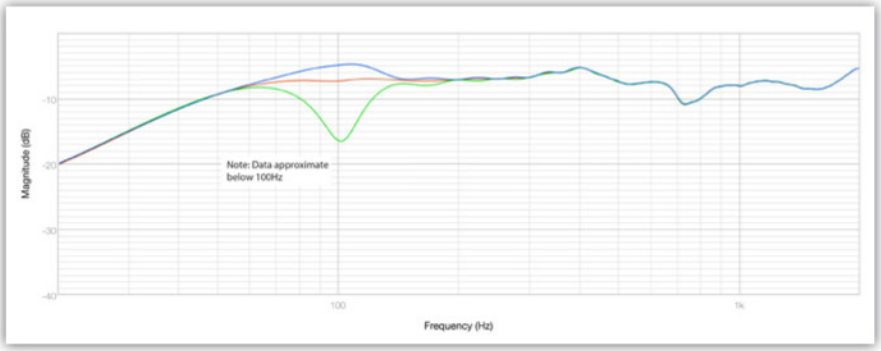


Diagram 5: Comparing the Aether in 'flat' mode (red) and with its EQ set to 100Hz, with a 3dB boost (blue) and 12dB cut (green).

» group delay at 40Hz of around 7ms. This is between a quarter and half of the figure I'd expect of a typical reflex-loaded monitor — although my recent experience of the Neumann KH150, with its 27ms group delay yet extremely satisfying bass, shows that minimising group delay isn't everything.

But what, I hear you ask, is the extravagant "horn loading" of the Aether reflex port about? I enquired of Greg at APS and he explained that it's designed to achieve three things. To begin with, it turns out that the horn visible at the rear of the Aether isn't the only one. There's another similarly generous horn component on the inner mouth of the port. The two horns firstly act together to maximise linear airflow volume and delay the point at which flow turbulence and noise occurs. Secondly, the horns modify the reflex port's coupling impedance at its entrance and exit to recover some of the output level lost through the high level of damping. And thirdly, the horns minimise any possibility of organ-pipe mode resonance along the length of the port.

"The Aether immediately revealed itself to be a very capable and well sorted monitor."

It seems to me, though, that there's another advantage: it enables such a low tuning frequency to be achieved without hitting the usual snag of running out of space in the enclosure for a port of the necessary length and diameter. The "APS horn bass reflex technology" in the Aether is an interesting approach to the implementation of reflex loading that I've genuinely not seen before — and things I've not seen before in speaker design are, these days, few and far between.

The final tyre-kicking measurement in Diagram 6 illustrates the behaviour of the Aether's room equalisation function. The red curve shows the Aether's frequency response in my studio room at the listening position. The two major low-frequency discontinuities potentially amenable to equalisation are the room-mode-induced peaks at 35Hz and 90Hz, but with only a single band of EQ available on the Aether I had to decide which one to take on. It's no contest, really: the 90Hz peak is much more likely to be significant in mix terms than that at 35Hz, so I dialled in a 6dB dip at 90Hz using the rear-panel controls. The green curve of Diagram 6 was the result, and it looks to me like an improvement. I also tried using the Aether LF equalisation function to suppress the 35Hz peak, but the EQ's roll-off is too gradual to provide any kind of targeted room correction.

Listening In

Moving on to playing some old favourite reference tracks and Pro Tools sessions, the Aether immediately revealed itself to be a very capable and well sorted monitor. My current reference is the Neumann KH150, but despite the Aether's contrasting design philosophies and technologies, it lives very much in a similar ballpark in terms of inherent capabilities as a mix tool. The first thing I think a monitor needs is a consistent and neutral tonal balance across its bandwidth, and the Aether has that nailed. It sounds slightly more upper-mid emphasised than the Neumann KH150, but the difference is marginal and there's no pervasive coloration associated with the characteristic, so it's easily learned. I also found the Aether a little bright at the top end, but knocking back the tweeter by 1.5dB using the rear-panel switch effectively brought the balance back into my subjectively preferred window. The Aether's simple room optimisation function also worked well to compensate for the 90Hz room mode at the listening position. It left the Aether sounding a little bass-light at other listening positions, however.

The Aether's tweeter is clear and detailed, but it is also one that just gets on with the job without particularly drawing attention to itself, and its integration with the bass/mid driver through the crossover region is clearly well managed. In theory, the relatively large-diameter bass/mid driver and the

ALTERNATIVES

In terms of price, the Aether sits in a highly competitive sector in the monitor market, with multiple exceptional products pitching for customers. Monitors such as the **Neumann KH150**, **Genelec 8340A**, **ADAM S2V** and **Dynaudio Core 7** would all be worth comparing to the Aether.

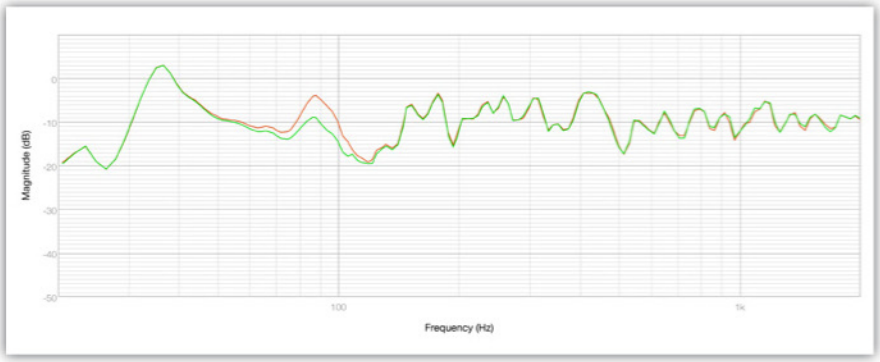


Diagram 6: The Aether's in-room response in 'flat' mode (red trace), and with its LF EQ set to -6dB at 90Hz (green).

relatively high crossover frequency of the Aether could result in a crossover dispersion discontinuity (narrowing bass/mid dispersion marrying with wide tweeter dispersion), but in practice there's no obvious signature of such a phenomenon.

The unusual low-frequency design of the Aether works well. The low-frequency bandwidth is very extended, and although I don't have the measurement facilities to confirm the Aether's published specification of -2dB at 32Hz, I wouldn't be surprised to find the reality not far off. And along with being extended in bandwidth, the Aether's bass sounds impressively dynamic and revealing of pitch and timing. It sounds more 'closed-box' in character than reflex. It's able to play at genuinely loud midfield monitoring levels without apparent strain or any obvious changes in character.

Moving up to the midrange, the Aether is highly revealing of mix detail and convincing in terms of balance. I'd have had no problem jumping straight into a mix with the Aether and feeling comfortable with the way it presents the all important voice band. Midrange stereo image focus is impressive too, with a fine corresponding ability to resolve reverb tails and the scale of acoustic spaces, whether real or plug-in generated.

To deploy an easy cliché, the Aether is something of a wolf in sheep's clothing. It's almost unbearably plain to look at, and apart from an unusual bass loading technique, it is, when stripped down to its fundamental elements, an entirely conventional two-way active speaker with an aesthetic that might have been born in the 1980s. However, to my ears it is clearly a monitor developed by somebody who genuinely knows their way around speaker design and how to optimise the parts to offer more than their sum. The Aether may look uninspiring, but it is genuinely able to do a really exceptional job of monitoring.

$ $3750 per pair.
W www.aps-company.com

KRK GoAux 4

Active Monitors

With room correction built in, KRK's highly portable GoAux 4 speakers are designed to deliver usable monitoring in any space.

PAUL WHITE

The KRK GoAux 4 active, two-way monitors are designed both for home studio and portable monitoring applications. Shipped in a tough a nylon carry case, along with a pair of adjustable-angle speaker platforms and an Auto ARC mic, plus the necessary cables, the whole package weighs in at just 4.3kg.

The Auto ARC mic is included because the GoAux 4 speakers have KRK's ARC or Automatic Room Correction built in. This is particularly useful for portable applications as it means you always ensure the best possible monitoring accuracy for the room you happen to be working in. The Auto ARC is a small omni mic with a long captive cable, and it has a clip that can attach to a mic stand or other support.

The speakers themselves measure 205 x 136 x 140mm and have distinctive KRK styling, with their four-inch woofers featuring the company's familiar yellow, woven-glass-aramid cone. At the HF end, and crossing over at 2.5kHz, is a one-inch, soft-dome tweeter. The dispersion of the speakers is specified as 120 degrees horizontal by 60 degrees vertical.

The cabinets are rear-ported and made from ABS plastic, with rounded corners and a reassuringly solid feel. The design is bi-amplified, with a total Class-D amplifier power of 100 Watts RMS. With a maximum SPL of 98.5dB (peak of 102dB), the speakers are loud enough for nearfield use and cover the frequency range 65Hz to 20kHz at the -3dB points (-10dB at 55Hz and 22kHz). That's pretty impressive given their compact size.

All the the power amps are located in one speaker's housing, and the second speaker is fed from there through a four-conductor link cable, so only one mains cable is required. Built-in LF and HF EQ settings (+2dB, -3dB or flat) provide scope for manual adjustment to suit the location of the speakers, though the built-in Auto ARC is able to measure the performance of the speakers in a particular space and then apply adaptive correction automatically.

There's no shortage of connection options, either. There's USB (which is class compliant, so no drivers are needed), a stereo mini-jack, RCA phono sockets and a stereo pair of quarter-inch, balanced TRS jacks. There's also Bluetooth capability, with a push-and-hold button routine to pair; a blue LED that blinks in pairing mode stays solidly lit once paired. A headphone mini-jack output is also available, and mutes the speakers when headphones are plugged in. The volume control includes a push switch for toggling the standby

KRK GoAux 4
$419

PROS

- Solid and detailed sound with a greater bass extension than might be expected.
- Simple ARC system for making the best of imperfect rooms.

CONS

- I really couldn't find anything to dislike given the size and cost of these speakers.

SUMMARY

A good solution for the smaller studio or for mobile recording.

mode on or off. The mains connection uses a small two-pin connector, so there's no possibility of ground-loop issues.

ARC At Thee

To use the ARC system, the mic is plugged into the mini-jack mic input on the front left of the master monitor, after which it should be set up at head height in your preferred listening location. Holding down the Auto ARC button on the rear of the left speaker for a few seconds starts the process, whereupon the speaker will generate a series of test tones over a period of a few minutes, during which time no other sound should be made and nothing should be moved. When complete you hear a brief low-frequency tone — and that's it, job done. The settings are retained until the next ARC measurement is made.

I have to admit to being unsure of what to expect from such a small pair of speakers, but the depth and clarity of sound from these monitors is seriously impressive. The low end is unexpectedly deep and solid, while the small woofer is also kind to voices and other midrange sounds. The dome tweeter handles the highs smoothly without losing detail, and if I hadn't been looking at the speakers when I first heard them, I could easily have assumed that the sound came from much larger speakers in conventional cabinets. You don't get the same maximum SPL as you might from full-sized monitors, but at a typical close monitoring position you certainly don't feel short-changed.

Before moving on to how the ARC system performs, it is important to appreciate what this type of room correction system can and can't do. Firstly, most room problems are what we call time-domain effects. For example, if you project sound into a space with a strong resonance at a particular frequency, the sound will continue to 'ring' for a short while after source has stopped. An EQ-based solution can tame the level of these resonances but can't correct this kind of 'overhang'. Sounds also reflect from hard surfaces, arriving at the ear later than the direct sound, but in a typical room these arrive soon enough that the ear/brain tends to merge them into a single sound. A typical ARC system works by identifying excessive peaks in the frequency response cause by resonances and reflection, then applying EQ to reduce the level being fed into the room at these problem frequencies. This can really help even out the sound at the monitoring position as long as the room acoustics aren't too bad to begin with.

However, acoustic problems don't just cause peaks in the response. They also cause dips, where certain frequencies are partially cancelled by reflected sound arriving later. Some of these dips can be very deep, so if a room correction system tried to correct them by boosting the dipped frequencies, the speakers would soon run out of headroom. For this reason most such systems apply little or no boost to frequency dips, and instead concentrate on taming peaks. As the human hearing system finds peaks more offensive than dips, this trade-off is worthwhile.

In a GoAux pair, the left speaker is designated as the 'master', and takes the mains power and audio signals. The amplified right-channel audio is then sent to the right speaker via a link cable.

Office Space

I tested the system in my office, which also doubles as a secondary studio for doing some reviews. Even without engaging the ARC system, the sound was immediately solid and impressive, even though my office has no acoustic treatment other than shelves of clutter and various boxes to diffuse the sound. Firing up the ARC process and then listening again, the sound was a little tighter and better focused, so the improvement was definitely worthwhile. Particularly impressive is the tightness of the bass — a dry kick for example comes over with very little evidence of overhang caused by the porting.

For mobile recording, where you often need to set up the speakers in odd locations, having ARC on board is clearly a big bonus, especially as it is so straightforward to use here. The included speaker platforms with their simple angle adjustment are also a big help when doing mobile work, as you never know what else will be available to use as speaker stands. Having a good-quality carry case included also makes it easy to move the monitoring system around.

In summary, then, whether you're working in a small home-studio room or doing mobile recording work, the KRK GoAux 4s do a fine job, delivering the kind of sound you'd expect from larger monitors where the only tradeoff is the maximum SPL. I'd certainly be happy to mix on them. And as stated earlier, there's still plenty of level in hand if you work at sensible nearfield monitoring volumes. These speakers quite literally punch above their weight.

$ $419 per pair.
W www.krkmusic.com

Toontrack EZkeys 2

Virtual Piano Instrument

Virtual piano, virtual pianist or virtual songwriting partner? Toontrack's EZkeys 2 attempts all three.

JOHN WALDEN

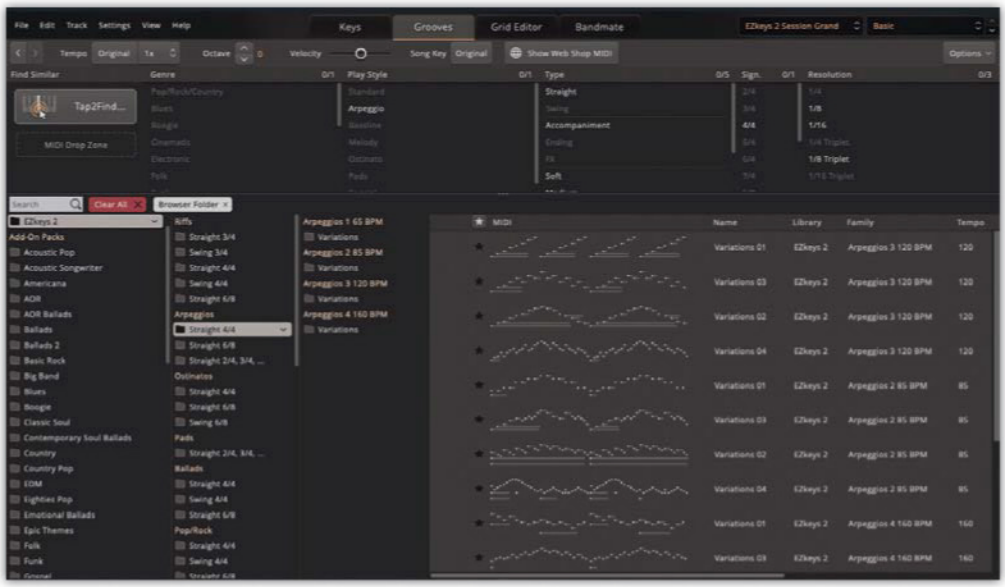
The arrival of Toontrack's EZkeys 2 has been highly anticipated. Ten years after the original release (see the January 2013 issue of *SOS*), and with Toontrack significantly upgrading the 'EZ' concept with more recent EZbass and EZdrummer releases, EZkeys was undoubtedly due a refresh.

In its original form, EZkeys served a number of functions. First, it provided a sample-based piano instrument that you could freely play like any other virtual piano. Second, EZkeys acted as a virtual pianist through its very flexible performance engine and the impressive (and expandable) collection of genre-specific MIDI patterns that could automatically adapt themselves to the user's chosen chord sequences. And, given the flexibility of the performance engine, and how you can experiment with chord and song structures within EZkey's Song Track, you could also argue that EZkeys offered a third role as a songwriting tool.

So just what have Toontrack done in the new release to push each of these three core functions forward? Let's lift the (piano) lid and find out...

Pure Piano

EZkeys 2 ships with a brand-new 'core' grand piano sample library clocking in at around 4GB. When I reviewed the original release in 2013, the engine and the sample content were sold separately, but this change brings EZkeys into line with the other EZ instrument products. The two original EZkeys 1 piano sample



The new core MIDI library includes some very useful riff, arpeggio, ostinato and pad patterns in addition to the genre-based content.

libraries — a grand piano and an upright piano — each contained just over 400MB of sample data. They were perfectly useable (and remain so), but sampling technology (and computer power) has moved on considerably in the last 10 years and the new EZkeys 2 core library reflects that.

Toontrack recorded a medium-sized Fazioli grand piano at Riksmixningsverket Studio in Stockholm to form the basis of the new instrument. A combination of high-quality Thuresson, Royer, AKG, Neumann and DeGeer microphones were used to capture both close and overhead/room sounds via a Neve console. Use of sustain, una corda and sostenuto pedals was included, as were noise elements from hammers and pedals. The EZkeys 2 engine lets you use and adjust all these elements as required.

From this underlying sample base, EZkeys 2 delivers a good stock of presets organized into three main groups: Standard, Effects and Pads. The Standard presets provide various versions of the conventional piano sounds, each tailored to provide a different character. This is achieved by blends of the various microphone positions and, behind the scenes, some gentle processing or EQ.

When auditioned side-by-side with the original EZkeys 1 grand piano, the differences in depth, detail and dynamics are very obvious; EZkeys 2 provides a very impressive and satisfying playing experience. Of course, there are solo piano libraries out there with much larger (+100GB) sample bases, but there is undoubtedly a point of diminishing returns in terms of the real-world playing experience. I'd suggest the actual acoustic piano sounds here will jump well above the quality bar required in the vast majority of musical contexts, even when the demands are high and the piano is heard totally in isolation; the EZkeys 2 core library is very impressive.

The two further preset categories provide more stylised/processed options. So, for example, the Effects category includes the suitably ambient Film Noir and tastefully degraded Lo-Fi Vinyl presets, while the Pads presets add a (mostly subtle) pad layer underneath the piano itself (for example, the Glory preset with a soft string layer beneath the piano). There is nothing too sonically groundbreaking here, but they sound great and are eminently usable.

It is worth noting that you also get a set of 'Raw Sounds' presets. These provide both stereo and multi-channel output versions of the core piano instrument and allow you to create your own blend of the various microphones each offers. Using the multi-output versions, you also have the option to apply your own processing choice to each of the microphone sets within your host DAW.

The redesigned Keys tab now follows the style of both EZbass and EZdrummer 3, with a smart graphical representation of the grand piano within its recording environment. When notes/chord are triggered, EZkeys applies its chord detection capabilities and the currently playing note(s) or chord are displayed in real time on the piano's virtual music stand. Like both other EZ instruments, located top right of the Keys page is the pop-open Effects panel. The options here are preset-specific but, amongst others, can include EQ, compression, reverb and noise, so you can easily adjust the presets further should you wish. It's slick, streamlined and very effective.

Toontrack will undoubtedly be offering further sound expansions for EZkeys 2 before too long but, if you already own expansions for EZkeys 1, these have now all had technical updates (available through the Toontrack Product Manager and now identified as EKX files) to make them fully compatible with EZkeys 2. This worked very smoothly on my review system.

MIDI Matters

EZkeys' second core function is as a virtual pianist, letting you quickly build an authentic piano performance that follows the chord sequence of your song. A key element of the instrument's ability to do that is the available MIDI library that is accessed via the Grooves page. This page has also been redesigned to provide a consistent look and workflow to the rest of the EZ instrument line and includes the Tap2Find option (familiar from both EZbass and EZdrummer), as well as the ability to find patterns by genre, play style, and other character tags, plus options to browse the folder and sub-folder catalogue. Patterns can be easily auditioned. »

Toontrack EZkeys 2

$179

PROS

- Core sample library sounds excellent.
- Revamped UI and feature set provides massive workflow improvements.
- Great value for money, either as a new purchase or an upgrade.
- Piano, pianist and songwriting assistant in one piece of software.

CONS

- None.

SUMMARY

In a virtual format, Toontrack's EZkeys 2 provides a great acoustic piano instrument, a session pianist and a songwriting assistant, all within a single piece of software. It's brilliant and will sell by the bucketload.

Grid (Un)lock

As with EZkeys 1, you can drag and drop your EZkeys 2 Song Track performance to a track within your DAW for editing (or playback by a different VI). However, as with the other EZ instruments, EZkeys now includes a Grid Editor page. This provides all the usual tools you would expect in a fully featured MIDI editor, with options to move, copy, add and delete notes, plus a range of additional tools that enhance the editing workflow.

For example, at the top of the editor display, you can toggle on the Humanize options and apply small timing/velocity changes to selected notes within a performance. The Timing button pops open further options for adding timing changes including adding swing. At the bottom of the display, you can make rapid adjustments to the velocity and dynamics of selected note/note groups or apply a degree of randomisation to both parameters. There are also options to view/edit lanes for note velocity and the use of the sustain, soft and sostenuto pedals.

The Grid Editor also offers the Highlight Keys feature. When open, it provides options for highlighting the Grid Editor's note lanes based upon the current scale/key, notes within the current chord, or notes within the scale of the current chord. And, as you can also toggle on a Scale Snap feature, unintentional 'duff' notes can be easily prevented. Overall, the piano-centric design of the Grid Editor makes it a very attractive alternative to a more generic MIDI editing environment within your DAW.

The new Grid Editor provides a comprehensive MIDI editing environment with a suite of tools – such as the Highlight Keys options shown here – that enhance your workflow.

The Grooves panel comes well populated. EZkeys 2 ships with a new collection of MIDI content containing over 1300 MIDI patterns that cover an expanded range of musical styles. These include ballads, pop/rock, hip-hop, soul/R&B, country, gospel, jazz, reggae, blues, funk, cinematic and boogie, and many of these offer a variety of straight and swung performances, as well as different time signatures. However, perhaps most welcome are the folders labelled Riffs, Arpeggios, Ostinatos and Pads. These house collections of more generic playing styles that are easily adaptable to a wide range of musical contexts. The arpeggios and ostinatos include options that are more sparsely played and so would easily sit under a vocal (for example) with minimal editing. The pads category obviously features sustained chords but make great starting points when initially constructing your chord sequence for a song section. If you already own EZkeys 1, or any of the existing MIDI expansion packs, these will work fine within EZkeys 2.

It's easy to underestimate the significance of the MIDI pattern library. It is, in essence, just a collection of MIDI clips, albeit carefully curated and expertly played. However, if EZkeys' 'performer' function is what drives your personal interest in the product, the MIDI pattern library is essentially the performance vocabulary; the more MIDI pattern content you have available, the wider the musical lexicon of your virtual pianist.

In that light, Toontrack's extensive collection of genre-specific MIDI expansion packs are well worth exploring. And, not to miss a trick, the Grooves page now provides the option to do just that, even for MIDI expansion packs you don't currently own. This is genuinely useful as it allows you to fully audition individual clips, giving a much better feel for the musical content of any pack. If you find something you like, one click will take you directly to Toontrack's web shop where you can add the pack to your shopping cart; very shrewd marketing but also a very slick process.

The Magic Of The Song Track

The Song Track has always been central to EZkeys' 'performer' and 'songwriting assistant' functions, so it's worth emphasising just how clever the Song Track concept is. As before, the user can add sequences of chord events to the Song Track either manually or by dragging and dropping patterns from the Grooves library. You can also record directly into the Song Track from a MIDI keyboard, and this includes the ability to merge a new recording with existing MIDI already held on the Song Track. Given the flexibility EZkeys then offers to customise the contents of the Song Track, this recording capability can be a useful feature regardless of your actual piano skills.

As shown in the nearby screen, entries in the Song Track are represented in a three-laned format comprising song parts (in the top lane; each song part is colour-coded/labelled and spans a group of chords), the chord events themselves (in the middle lane), and a representation of the MIDI performance currently associated with the chord sequence (in the bottom lane).

The feature set of the chord wheel has been expanded and the workflow improved.

The thing that's always made the

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The Replace MIDI function is both clever and powerful, allowing you to easily audition your chord progressions with different playing styles.

Song Track so powerful (and, yes, somewhat magical) is EZkeys' ability to adapt an existing MIDI performance to changes you make to the actual chord sequence, or to apply a different performance to the same chord sequence... or, indeed, some combination of both of these things. These creative possibilities now come with both a better workflow (courtesy of the improved EZ UI) and some significant additional options.

If you wish to change a chord, you simply double-click on the chord label within the Song Track's middle lane (or click on the Edit Chord button), which opens the familiar chord wheel. If you then select a new chord within the wheel, the MIDI performance within the Song Track is adjusted to match the new chord while retaining the existing playing style.

While EZkeys 2 can automatically make pianist-style choices about chord inversions to produce a smooth performance, small changes to the chord wheel's control layout make it easier to adjust the chord inversion manually (you hear an audition of the chord as you make changes) or to define an alternative bass note. The chord wheel now also has a further ring that can be popped open on its right side to access additional chord variants. And, if you want even more control over the combination of notes that form your chord, clicking the small keyboard icon located top left of the chord wheel opens a keyboard view. Here you can simply select/deselect notes as required (the relevant scale degrees are highlighted on the graphic making it easy to add sixth, seventh or ninth notes, for example) and you can also select the bass note to play. Whether it's simple triads or harmonically sophisticated chords for jazz, the chord wheel now gets you there with ease.

Alternatively, for any given set of chords, you can experiment with different performance styles; simply select the song section you wish to experiment with (via the Song Track's topmost lane) and then activate the Replace MIDI button. This then opens a MIDI pattern browser panel in the upper part of the UI. When you select a MIDI pattern here, its performance style temporarily replaces the MIDI within your selected Song Track song section but adapts it to fit the existing chord sequence. This is super-easy to use and makes it possible to experiment with any manner of stylistic options for your song's chord sequence. And, if you find something you like, you simply click on the Save Changes button and the new MIDI performance permanently replaces the old one within the Song Track itself.

Usefully, EZkeys' Edit menu does offer undo/redo if you suddenly wish to undo such a change. However, the option to have multiple versions of your Song Track, and to create a duplicate of an existing version, means that you can easily experiment with these sorts of changes without any fear of losing your original. The Song Track also allows you to create tempo, time signature and key signature changes within a project (via the menu button located within the Sig/Tempo/Key panel at the base of the UI). When used within a DAW host, EZkeys 2 follows your host's tempo including any tempo changes. However, if your DAW project does contain time signature changes, you will still need to recreate those same changes within the Song Track. Unless you use lots of time signature changes (any math or prog rock fans out there?), this is a relatively modest inconvenience.

The Song Track also gets a further very significant new feature: Edit Play Style (shown in the screen at the start of this review). This functionality — seen in both EZbass and EZdrummer — provides the user with a range of options for adjusting the MIDI performance data within the Song Track. The available controls appear in a strip when you engage the Edit Play Style button. Adjustments made here will be applied to whatever is selected within the Song Track so they can be used on an individual chord, a song section, or multiple song sections, as required.

The purpose of the Octave, Velocity and Length controls are pretty obvious, but it's the Amount control that's particularly clever. This can be used to either subtract or add notes within the selected portion of the performance, making the playing style sparser or busier. Behind the scenes, EZkeys attempts to do this in an intelligent, pianist-like fashion. The whole Edit Play Style concept is excellent so, if you initially start out with a sequence built from identical copies of the same MIDI library pattern, you can easily add touches of performance variety to each copy while retaining the overall style and dynamics of the playing.

Join The Band

While EZkeys makes an impressive virtual pianist, as Toontrack have now also added the Bandmate page, it can also join your virtual EZ band. As seen in EZbass (where it's spread across the Drums & Bass and Audio Tracker pages) and EZdrummer 3 (where it first appeared as Bandmate), this feature allows you to drag and drop either audio or MIDI performances into EZkeys and the instrument then suggests MIDI patterns from its own library that might provide a good match. You can customise the genre used for the selection of matches shown if you are looking for something in a specific musical style. And,

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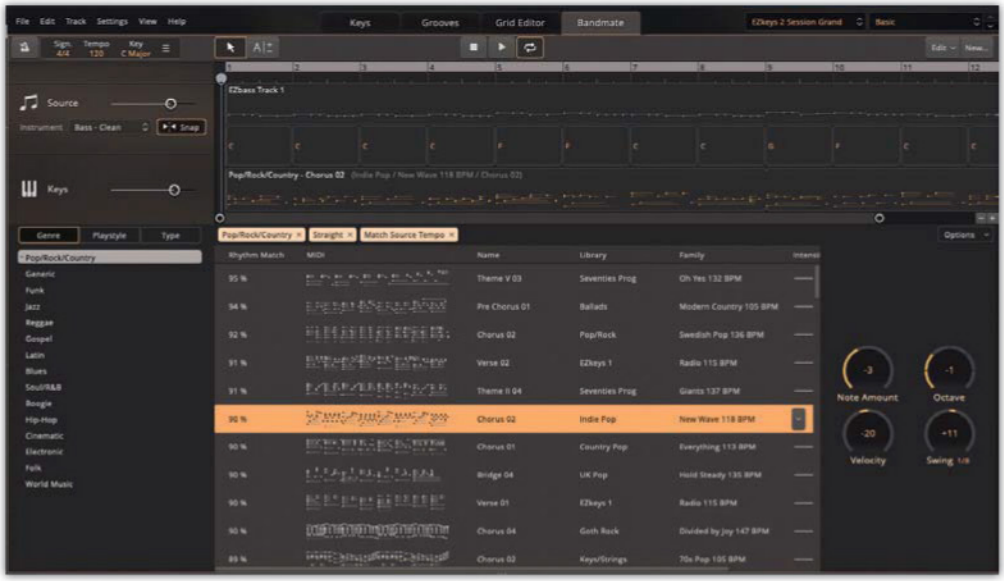


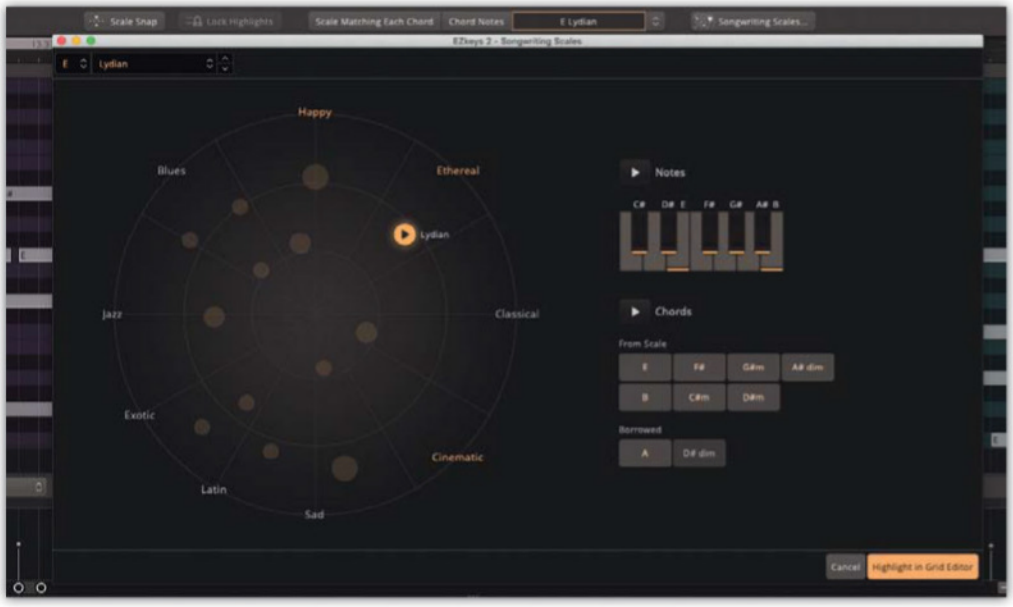
The Bandmate page lets your virtual pianist interact with your other EZ band members or audio/MIDI performances from other sources.

rather wonderfully, once you highlight a MIDI pattern and audition it, the Note Amount, Octave, Velocity and Swing controls let you customise it on the fly to suit your needs.

If it's an EZbass sequence you drag in, then the associated chord sequence will also be imported and, as you then audition the possible matches, the EZkeys patterns are mapped to the EZbass chord changes. It's a brilliant way to get started on a piano part if a bass line was your initial musical inspiration. Equally, note that you can also drag and drop EZkeys patterns into the equivalent Bandmate-style functions within either EZbass or EZdrummer and the same sort of pattern matching process can quickly find you something to work with. Drag and drop audio — for example, a DI'ed guitar recording — into EZkeys' Bandmate and it will make a decent first estimate at the chords and rhythmic content before generating its list of pattern matches. It's not foolproof, but it's almost always a decent starting point.

The Bandmate feature — and how it integrates across the EZ world — is a heck of a trick. The speed with which you can get your three-piece EZ virtual band playing in sync is remarkable. It does, however, provide multiple workflow possibilities depending upon where you like to start writing (drums, bass, piano or an audio performance?). New users may need to devote a little time to find their personal preferences here. Bandmate also highlights the value of having well-stocked MIDI groove libraries for each of your EZ instruments. This represents the catalogue of performance styles that Bandmate can call upon to find performance fits between the various 'bandmates'.

I really do hope that Toontrack continue to develop how these three virtual musicians interact with each other. For example, I wonder whether we might eventually see the option to automatically update changes made to the chord sequence in EZkeys' Song Track to the Song Track in EZbass (or *vice versa*)? That said, once you find a workflow that suits you, the current Bandmate experience is very impressive and it's great to now see EZkeys more closely integrated into that.



The Grid Editor's Songwriting Scales feature can encourage you to write in musical styles or scales you might not otherwise explore.

Computer Collaborator

The Highlight Keys feature of the new Grid Editor (see box) includes another boost to EZkeys' songwriting assistant credentials, namely Songwriting Scales. This opens as a panel with a large circular graphic around which a combination of genres/moods are represented by smaller circle icons. If you select one of these, it triggers audio examples of the selected key/scale combination and its musical flavour. The right side of the window shows the associated notes and chords, and these can also be auditioned. The Highlight In Grid Editor button then does exactly what it suggests and, if you enable the Scale Snap button, any subsequent note pitch editing will snap to this new scale. Even if music theory is not your thing, the Songwriting Scales window could easily encourage you to create melodies and chord sequences outside your usual comfort zone.

However, perhaps the headline songwriting feature is the new Suggest Chords option, accessed by clicking a button at the top of the Song Track. The upper section of the UI then shows the sequence of chords in your currently

The Song Track's new Suggest Chords feature is a brilliant way to explore and experiment with chord progressions.

selected song section (up to a maximum sequence of 16 chord changes) along the top row and, beneath each of these chords, a column of five chords that might work as alternatives for each step. EZkeys makes these chord suggestions on a genre basis, and you can specify the genre from a drop-down list of options.

You can then simply experiment with changing the chords, either individually or for multiple steps within the sequence. You can also lock a chord to prevent it being changed. As you pick an alternative chord, it is substituted into the song section within the Song Track, and you can audition the individual chord and listen to it in the context of the chord sequence. As you change one chord, the chord suggestions for the subsequent steps are automatically updated to provide new choices suitable for the change just made. If you wish to return to the original chord within a step, simply click on the original chord within the top row. And, if you simply want Suggest Chords to cough up full or partial chord sequence suggestions for you, then the Change Chords button will use its built-in musical knowledge and keep doing just that until you hear something that you like.

This whole process is both simple and intuitive. You can start with a MIDI pattern from the EZkeys library or from a simple eight-bar sequence generated via the Add Standard Groove button (which places a song section on the Song Track filled with eight one-bar instances of the root chord of the key). You then simply experiment while EZkeys guides you with suggestions based upon the key/genre you have chosen. Yes, it's still up to you to make the final choices (good or bad) but, whatever your level of music theory knowledge, this is a great playground. And, once you are happy with your chord selections, simply closing the Suggest Chord window locks them in to your selected Song Track section. Suggest Chords is a truly inspiring tool for exploring and generating chord sequence ideas. And with it, Toontrack have taken EZkeys' capabilities as a songwriting assistant to a completely new level.

Going For A Song?

As indicated at the start of this review, EZkeys is a virtual instrument that can serve as a source of acoustic piano sounds, as a virtual pianist (and member of a virtual band) and as a songwriting assistant/collaborator. As a source for acoustic piano sounds, in terms of sonic details, dynamics and playability, EZkeys 2 represents is a big step forward compared to the original and, for the majority of users, it will more than pass the required quality bar, even used in isolation.

As a virtual pianist, the basic EZkeys concept remains the same but the new UI, and its many new features for customising the performance, makes for a more powerful and more efficient workflow. The new core MIDI library is also excellent. As a brief aside, here in EZ world, we now have a very impressive virtual band of drums, bass and piano... so it does beg the question as to if/when we might get to see EZguitar? I can't believe Toontrack have not considered this possibility in detail, but I also don't believe they would do it unless they felt it would match the standard of the existing EZ instruments.

You could argue that within the original EZkeys, the 'songwriting assistant' was a more minor component. That's no longer the case. Concepts such as the Songwriting Scales (surely something Toontrack will develop further) and the brilliant Suggest Chords feature bring this role into the spotlight.

And, while all three of these individual 'parts' of EZkeys have been significantly improved, it's the sum of those parts — the ease with which they all work together to make a coherent whole — that is truly impressive. A great-sounding acoustic piano, a powerful virtual session pianist, and a songwriting collaborator that will happily throw endless chord sequence possibilities at you to see what inspires. Given what's on offer, the asking price represents exceptionally good value for money and the upgrade an absolute bargain, especially if you fully exploit the second and third of these roles. Toontrack have done it again — EZkeys 2 is simply brilliant.

$ $179. Upgrade from EZkeys 1 $99.
W www.toontrack.com

Zivix Jamstik Classic

MIDI Guitar

DAVE LOCKWOOD

The launch of the original Jamstik Studio marked a real advance in the world of pitch-to-MIDI guitars — and there's now an alternative model for guitarists who prefer to play something more familiar.

When I reviewed Zivix's Jamstik Studio MIDI Guitar (www.soundonsound.com/reviews/zivix-jamstik-studio), I wrote that "if there is ever a Mark II of the Jamstik Studio, I hope it has some easier-to-use tuners." Well, here it is! The new Jamstik Classic mounts Zivix's innovative and highly effective pitch-to-MIDI conversion system into a Strat-like design, complete with a conventional headstock and tuners.

It is not that the previous incarnation didn't work — I must admit I've very much got used to tuning with a hex key at the bridge over the last two years — but with MIDI guitars you do tend to check and re-tune a lot, and needing a tool for this basic function is just not as convenient as reaching up and being able to tune with your fingers. Of course, there are also some people who simply don't get on well with headless designs: if you are already an experienced player of conventional guitars, it does slightly disturb your inherent sense of where you are on the neck with your fretting hand, for a while at least. Although the original Zivix model, which retains the name Jamstik Studio, had a conventional 25.5-inch scale length, it featured a scaled-down body size that also imposed a different right-hand position compared to a 'normal' guitar, so the full-size alder body of the new Classic will be welcome in that respect, too. The neck on the Classic model is a 25.5-inch scale-length, 22-fret, roasted-maple, bolt-on design, with a comfortable 42mm width at the nut.

Although this is a Strat-like guitar, there's no floating trem — on balance, that's probably wise in the MIDI-guitar world, where stable tuning is so important — but the tuners are nevertheless the locking type normally associated with a trem bridge (yes, I know it should be vibrato not tremolo, but I'm not sure I can personally undo 70 years of misusage...). Some might argue that locking tuners offer inherently more reliable tuning anyway, although I'm not sure I altogether agree. In fact, if you follow the locking-tuner 'best practice' of using the minimum wrap around the string post, you end up limiting the break angle over the nut of the low E and A strings. Two double string trees take care of the break angle for the other four strings, and, of course, there's nothing to stop you using

a ‘sensible’ number of wraps on the E and A to get the break angle you want.

Setup Matters For MIDI

The familiar double set screws of the six chunky, chromed bridge saddles adjust the action in the conventional way using the supplied hex wrench, and the brass nut is adjustable for height, too, if necessary. The truss rod adjusts from the headstock end, with the neck responding very quickly to small tweaks. This may all seem like it has nothing to do with the all-important MIDI side of things here, but the fact is you need a decent basic setup for the guitar to avoid compromising MIDI performance, no matter how good the pitch-conversion technology used. A low action obviously helps you to play cleanly, but fret buzz creates pitch ambiguity for the detection system, so it’s all about finding the best compromise.

The pickups on the Classic consist of a chrome-covered bridge humbucker and two ceramic-magnet Strat-like single coils, with the Zivix system’s narrow hex pickup nestled in behind the humbucker. There’s a Strat-like single volume control, and two tone controls, the latter assigned to the two single-coils, leaving the bridge humbucker without one. I suspect some players might prefer the second tone control attached to the bridge pickup, but it’s an easy fix for anyone competent with a soldering iron. A five-way pickup selector switch offers all the individual selections and parallel combinations except neck and bridge. The middle pickup has a reverse-wind/reverse-polarity spec, to make it in-phase and hum cancelling when combined with the neck pickup, but there is no auto-coil tap on the bridge humbucker to keep it hum cancelling when used with the middle pickup. A humbucker and a single coil in parallel doesn’t ‘quack’ quite like two single coils, which means you miss out on one of the classic Strat tonalities. The humbucker is also slightly further from the bridge than on a conventional ‘super Strat’ due to the presence of the hex pickup. Tonally, I don’t find that a problem, and I actually think a humbucker sounds better placed there on a 25.5-inch scale length.

I’m sure that classic conventional guitar tones will be very much a secondary consideration to purchasers of this instrument, however: it’s the MIDI performance that we are really interested in. An edge-mounted USB-C port is used for both MIDI output and charging the onboard battery, with a 3.5mm mini-TRS MIDI output, too, and wireless MIDI connectivity using BLE (Bluetooth Low Energy) also onboard. I always prefer a wired connection for minimum latency MIDI recording, but the Bluetooth connection facilitates the use of the new »

The original Jamstik Studio model (right) next to the Strat-like Jamstik Classic, which is available in Vintage Cream or Onyx Black.

Zivix Jamstik Classic
$999

PROS
- Fast, accurate MIDI output from a real guitar.
- Outputs MIDI directly.
- Jamstik Control app makes settings easily accessible without having to go into the Creator instrument.
- AAX Pro Tools version of Creator now available.

CONS
- More physical MIDI controllers would be advantageous for live performance.

SUMMARY
Pitch-to-MIDI combined with some DSP ‘intelligence’ has resulted in a major improvement in MIDI guitar performance, and the integration of Zivix’s technology into a more familiar guitar format can only make it more accessible to a greater number of players.

» Jamstik Control app on mobile devices. You can (and I do) use both together: the Bluetooth for settings in the app and the wired connection for MIDI. This allows you to access all the performance and setup parameters of the MIDI side of the instrument on a conveniently placed tablet or phone, without having the Jamstik Creator app open and occupying a chunk of your DAW's screen space. A lot of the work that I do requires the sounds of 'conventional' instruments, albeit in virtual form, and that's not an area in which the sound library of the Jamstik Creator app is particularly strong, I feel, even with some new libraries available. Previously, working in Logic Pro, I would have to fire up Creator solely to flip between 'no-pitch-bend' mode for a piano part, and then activate pitch-bend and perhaps set the pitch-bend range for a fretless bass part, for example, but now I can just do that from the app on a tablet or phone.

Both the Creator software and the new Control app also offer a Transcription Mode, specifically for use with music notation software. This is designed to maximise accuracy by disabling some of the note detection features and also increases latency by some 30-40 milliseconds, so obviously not intended for real-time performance.

Jamstik Creator Instrument

The virtual instrument (VST and now also AAX Pro Tools-compatible) within Jamstik Creator does, however, make effective use of MPE mode (see the MPE boxout), an extension to the MIDI spec that allows for much deeper control and interaction with potentially every note than is possible with just channel-based MIDI controller data. MIDI guitar systems were, from the very start, often configured with each string being on its own MIDI channel so that each could have its own pitch-bend data, rather than a single channel-based pitch-bend parameter that would, of course, change all strings' MIDI output at the same time. But the MPE spec allows for real-time control data to be applied on a per-note basis. With a Jamstik in MPE mode and with String Envelope activated, MIDI CC11 (Expression) or CC7 (Volume) data is generated based on each note's string amplitude, tracking its envelope and using that data to influence synth sounds. More String Envelope control data options are available, including Channel Pressure (Aftertouch) and CC71 (usually Resonance), and can make for a more expressive playing experience, along the lines of 'multidimensional' controllers like the ROLI Seaboard, Roger Linn's LinnStrument and other dedicated controllers that permit meaningful physical interaction with MIDI notes as they are sounding.

The new Jamstik Control app for mobile devices gives you access to the settings and performance parameters independently of the Jamstik Creator virtual instrument.

Generating the MPE control data is only half the picture, of course: the receiving synth has to be set up to know what to do with it, and the majority of virtual instruments included with DAWs certainly aren't yet 'MPE ready'. A deluge of MPE data will often prevent a single-channel virtual instrument from working at all, so if you are having problems with a Jamstik connected to a DAW, the first thing to check is that you haven't inadvertently left it in MPE mode. It is easily done as the Creator app always prompts you to enter MPE mode for tuning and setup. Many of the sounds in the eight new sound libraries available to purchase for Jamstik Creator, some of them created in collaboration with Brooklyn, NYC-based Samples From Mars, seem to make effective use of the expressive possibilities of MPE and are well worth exploring.

Standalone Use

Although I'm sure the primary application for Jamstik instruments, among *Sound On Sound* readers at least, will involve connection to a DAW, Jamstiks

Understanding MPE

Officially, MPE stands for 'Multidimensional Polyphonic Expression', although pretty much everyone, including seemingly the MMA (MIDI Manufacturers Association), seems to refer to it as MIDI Polyphonic Expression. The fundamentals of MIDI data haven't changed since their adoption in 1982, of course, allowing for note information to be generated from a control surface, usually a keyboard, and sent to virtual or hardware instruments, or recorded in a DAW software application. The basic data says which notes have been played and for how long (and on which channel), whilst additional data describes other parameters, such as pitch-bend, sustain pedal use, or modulation application, and applies it to all notes being played on the same MIDI channel.

MPE, by contrast, sends its additional control data on a per-note basis, allowing perhaps one or two notes in a chord to be emphasised with modulation or a filter opening up, while the others are unaffected. This is far more like interacting with a physical instrument than playing sounds from a conventional MIDI controller like a keyboard, so long as there is some logical connection between the player's actions and what happens to the sound.

MPE-compatible controllers tend to have dedicated virtual instruments already set up to make use of the data so you can just get on with the business of playing and exploring the additional expressive possibilities. At present, however, I'm not aware of any MPE-ready virtual instruments included within DAWs, nor is the editing of MPE data well catered for in every DAW, and the ability to easily edit is, of course, one of the greatest assets of the MIDI format. I'm sure makers of expressive controllers will be keeping a close eye on software developments in this area, which are surely bound to come.

The Jamstik Creator app has had a number of enhancements since our original review and now has access to many more libraries of sounds.

output MIDI directly and are therefore capable of standalone operation, either connecting directly to a hardware MIDI synth or to the Jamstik Creator soft synth for performance. In my review of the original model, I felt that it would have been far more useful in standalone mode if it had hardware control of some of the basic settings on board, such as multi- or single-channel operation, pitch-bend on/off, and octave selection. These are now all well taken care of in the Control app, making it ideal for studio work, but I'm not sure I'd want to be dependent on an app for those settings in a live performance scenario. Along the same lines, I'd happily dump the second tone control in favour of a MIDI CC7 (Volume) generator. My venerable old Casio MG510 didn't do MIDI half as well as the Jamstik Studio, but it did have all of those parameters on hardware switches, and a built-in tuner, too. Maybe some considerations for a future Mark III? One new thing we do have now, however, is a double-tap facility on the power-on switch to flip between single-channel and MPE operation. It would be great if that were a user-definable function in the control app (personally, I'd prefer it as pitch-bend on or off).

Still The Best?

I described the original Jamstik Studio as "one of the best guitar-to-MIDI instruments I have used" and the time I have spent with it since then has only increased my appreciation of its capabilities. Several firmware updates have managed to avoid breaking anything, so it still provides cleaner, more reliable MIDI than any other pitch-to-MIDI based system I have experienced. The MIDI system in the Classic is exactly the same as the one in the original Studio model, confirmed by the measured times from pick transient to the start of the converted MIDI note being almost exactly the same on both models. I say almost because, as in my original review analysis, it doesn't always seem to be entirely consistent. The top E at 5-6 ms has too narrow a window to ever notice anything different, but I'm still both baffled and intrigued by the apparent ability of the low E, with a measured time in isolation of around 25 to 30 ms, to sometimes seem able to place notes earlier than it should with that conversion time. Have I got magically better at consistently compensating for latency on the lower strings? I very much doubt it. I have asked Zivix for any insight but, as before, they are understandably shy of revealing the details of their proprietary technology other than to say that it is "new and developed entirely in-house".

The original Jamstik Studio remains current and, as the name perhaps suggests, maybe that's the one for people who do their work sitting in front of a screen and keyboard all day; the smaller body and headless neck are actually quite useful in that context. The new Jamstik Classic, with its more traditional and familiar guitar-like persona, will certainly make it easier for players to transition into the world of MIDI guitar for the first time, as well as completely eliminating the minor irritation of tuning the strings at the bridge, and could certainly be seen as better suited to those who might want to use both MIDI and conventional guitar sounds from one instrument in a live performance setting.

It takes a fair amount of faith and dedication to continue to actively innovate and develop instruments for the narrow and highly specialised world of MIDI guitars, but given the growing interest in more expressive MIDI instruments and need for alternative controllers to exploit them, it's not too hard to imagine that Zivix's persistence might have put them in exactly the right place at the right time.

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Sonible true:balance

Reference Metering Plug-in

JOHN WALDEN

When it comes to achieving a balanced frequency distribution in your mix, Sonible's true:balance metering plug-in can let your eyes support your ears.

Comparing the overall frequency balance of your mix with that of well-regarded material in the same style/genre remains an important quality control step, even for seasoned engineers in the very best studios. But it's arguably a more critical step for those working in project or home studios, where the monitoring environment is usually further from perfection. Your ears must be your primary guide in this process, of course, but a visual comparison of the frequency distribution of your track with that of those you're referencing can offer helpful confirmation that what your ears are telling you is correct.

Plenty of plug-ins offer a visual indication of the frequency balance, ranging from the frequency analysers found in so many EQ plug-ins to more specialised software designed specifically to make this comparative task easy. In the latter category, Sonible now offer the modestly priced true:balance, which can be bought on its own, or in a bundle with the company's true:level loudness and dynamics meter.

Polished Visuals

As with other Sonible plug-ins, true:balance's GUI is a stylish affair. The central portion is dominated by a frequency display that shows three key components: a real-time frequency analyser for your audio input (in solid green); a time-averaged version of the same signal (the pale green line); and an envelope (in grey) showing the frequency response of your chosen reference audio. This enables a direct visual comparison with your own audio.

At the top of the screen you can choose the source of your reference. This can be one of the genre-specific presets supplied by Sonible (on the left) or up to eight reference tracks of your own choice, which you can load into the available slots on the right. Beneath this, you get numerical and visual feedback indicating how closely the frequency response of your own audio matches that of the selected reference across three broad frequency zones (lows, mids and highs). If you engage the Balance Check button, you also get text prompts overlaid on the graphical display that suggest specific adjustments to your track's EQ balance that you might consider if your aim is to more closely match your reference.

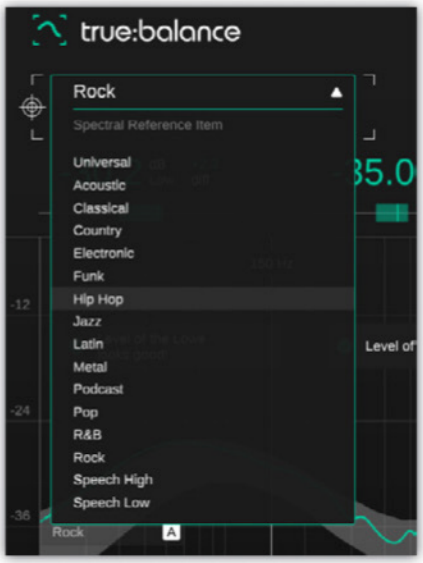
Below the frequency display, you get similar numerical/visual feedback on the stereo width of the same three frequency bands and, again, engaging a Mono Check button will generate text-based advice for you to consider. For example, it can help you identify whether your audio might suffer when monitored through a mono playback system. If you drag on the level scale on the left of the display, you can scroll the range of values shown to best fit your source track's input signal level range. Finally, on the right edge of the display, there is an output meter showing peak and RMS values.

By default, the low/mid and mid/high crossover frequencies are 150Hz and

4kHz respectively, which seem perfectly choices sensible to me, but if you wish you can change these by clicking and dragging on their labels on the display. At the base of the frequency display, you can also adjust the speed and resolution settings for the time-averaged (light green) frequency curve of your own audio.

Good References

Given true:balance's primary function, the options for choosing reference sources are an important feature. The supplied genre-based presets cover a good selection of musical styles and, as might be expected, the frequency distribution envelope for each style is somewhat different. These are presumably based upon some pretty sizeable databases of commercial material within each of the named genres. But if you have more specific targets in mind, you can easily load your own choices into the eight user Reference Track slots; I had no problems loading WAV, AIFF or MP3 files. Usefully, as you hover the mouse over a specific slot, you get a reminder of the file name for the reference source in that slot. Even more usefully, you can toggle individual slots on/off: the reference envelope adjusts itself to represent an average of the active slots, and you also see a colour-coded line within the envelope that represents the time-average for each active reference individually. I found this flexibility very useful: not only is it helpful if you wish to match your own track with a specific selection of tracks in your



Sonible provide a good selection of genre-based reference presets.

Sonible true:balance

€39

PROS

- Good visual feedback for comparing the frequency balance of your own tracks to suitable references.
- Hints may be useful for inexperienced amateurs.
- Simple to use.
- Very competitively priced.

CONS

- None really, though not everyone will find the hints useful.

SUMMARY

Sonible's true:balance provides very useful visual feedback on the frequency balance of your mix or master by comparing it to reference sources, and can suggest tweaks too.

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» target genre, but also for when you're trying to ensure tonal consistency across an EP or album project.

The frequency display also includes an intriguing A (Reference Anchor) button. With this engaged, the reference envelope will automatically adjust itself on the vertical dB-based scale to follow the signal level of your source audio. With this disengaged, you can drag the anchor up/down. This opens up some additional possibilities. For example, in addition to your visual frequency matching, you could visually level-match multiple tracks in an album project to a specific reference curve/level combination. This could result in a more consistent batch of mixes to work with when mastering. (That said, this functionality doesn't replace a dedicated loudness meter.)

Seeing Is Believing?

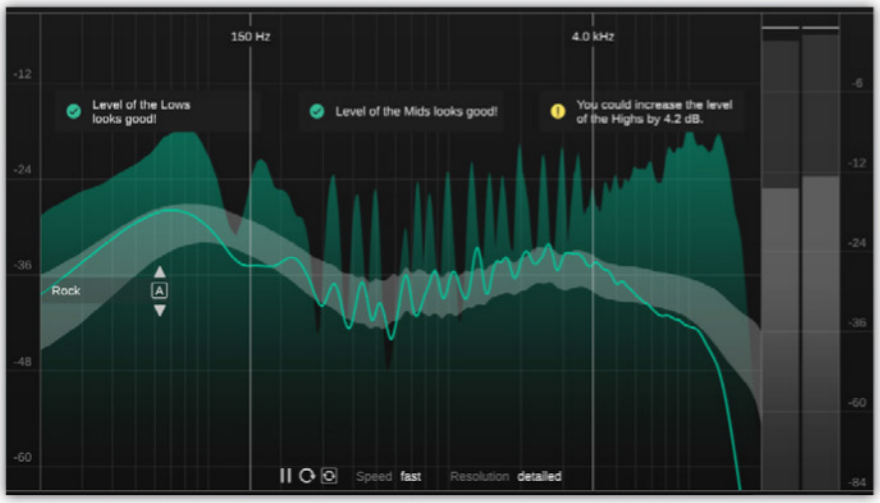
Used in my own DAW and audio editor of choice (Cubase Pro 12 and WaveLab Pro 11, respectively), true:balance behaved flawlessly during the review period, so that really just leaves the question of how useful this plug-in is — and to answer that question I need to tell you a little about my studio setup, and to discuss the software I've been using in this metering role to date.

I have a dedicated space for my music production work but, as with so many project/personal/home studio rooms, despite good use of acoustic treatment and a dollop of room correction software, that space has its limitations. Over the last five-plus years, I've used iZotope's Tonal Balance Control plug-in, which is true:balance's most obvious competitor, in pretty much every mixing or mastering project I've worked on — it has been my visual safety net, and I'm sure it has helped me refine my mix balance and achieve more consistent results.

Sonible's true:balance can easily perform the same role — so if you also have doubts about what you're hearing in your own mixing space, and could benefit from a bit of additional reassurance, I'd encourage you to try either of these plug-ins.

Test Spec

Cubase Pro 12.0.60. iMac running OS X 10.15.4, 3.5GHz Intel Quad Core i7, 32GB RAM.



The Speed and Resolution options allow you to adjust the visual detail of the graphical display.

An obvious question, then, is whether true:balance could replace Tonal Balance Control 2 in my own workflow, and the simple answer is that I'd happily use either. In terms of their core functionality (the visual feedback, and ability to compare the frequency balance of your own track to some form of reference track), they are very similar in operation, and provide similar information. Even the preset reference curves cover similar genres and seem suitably consistent, although there are some modest scaling differences between the two visual displays.

> "Metering plug-ins may not have the 'buy me now!' allure of a new virtual synth... but they can play a hugely important role in your studio."

The Balance Check and Mono Check features of true:balance can be useful, although perhaps that's more the case for less experienced engineers, since the suggestion that 'you could reduce the level of your lows by 1.5dB' doesn't really offer you much more insight than that you can glean from the curve, envelope and numerical displays. I like the way in which the user section of true:balance's reference track feature is implemented — it's both easy to use and visually useful — though it lacks the Tonal Balance Control 2 feature whereby you can assemble a whole list of tracks into a folder on your hard drive and have the plug-in construct an average reference envelope from them.

On Balance

If you're already a Tonal Balance Control 2 user, then, true:balance might well fall into the 'nice, but not essential' category. But if you don't currently own either, and are looking for this kind of metering option, then there's plenty to commend both plug-ins. Despite some feature differences, when it comes to their core function either is more than capable of doing the job. Both have fully functional time-limited trial versions if you want to compare them for yourself, so it's at this point that price might become a deciding factor. As a stand-alone purchase, true:balance is the more affordable option, and is even better value when purchased as a bundle with true:level. That's likely to be a very persuasive factor for many potential purchasers, though it's worth noting that Tonal Balance Control 2 is also included in some of iZotope's bundles.

Metering plug-ins may not have the 'buy me now!' allure of a new virtual synth, sample library or effects processor, but as with good monitoring and acoustic treatment they can play a hugely important role in your studio. It's so important that you can have confidence in the technical aspects of your mix as it leaves the confines of your own studio space, and Sonible's true:balance can help to give you that peace of mind — whether the song itself is good, bad or even downright ugly, at least you can be reassured that its tonal balance is in the right ball-park for your chosen genre!

$ true:balance €39 (about $43). Metering Bundle (with true:level) €69 ($75).
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Line 6 Catalyst 60

Modelling Guitar Amp

Some modelling amps can scare off potential purchasers by being too complicated, and with that audience in mind Line 6 have taken a very streamlined approach with their Catalyst range, while still managing to provide all the features needed for stage, studio and home practice. There are three models in the series, and we were sent the smallest in the range, the Catalyst 60, whose price is lower than that of most multi-effects pedals! There are six Line 6 Original Amp Designs that cover all the essentials from pristine clean to uber-filth, with some nice crunch and classic rock sounds along the way. You also get a generous selection of effects, which are drawn from the company's HX and Legacy effects libraries, plus a basic tuner.

Equipped with a single 12-inch speaker, the MDF cabinet is open-backed, and its black leatherette covering and dark speaker grille give it a very traditional appearance. At just under 11.8kg, measuring 444 x 503 x 262mm and with a top-mounted strap handle, it's easy to move around, and an optional cover is available.

The gain, volume and tone controls will be instantly familiar to guitar players, and the control panel is arranged to offer two channels comprising A and B presets plus a manual setting. There are also six banks of channel presets that you can access from the front, each bank holding a Channel A and Channel B setting. A rotary selector switch is used to choose presets, the amplifier type or effect type, and there's a switchable Boost with its own level control. There are 24 onboard effects, an effects loop, a power amp input, a power level control, and a balanced XLR output with cabinet emulation and switchable ground lift. The rear panel has the usual IEC mains inlet and power switch, and there's also an amplifier power control that can be set to mute, 0.5W, half power or full power.

There's also a USB-B port on the rear that can be used to connect a Mac/Windows computer running Catalyst Edit software for deeper effects editing, or to allow the amp to function as a four-channel, class-compliant audio interface with 24-bit resolution and supporting sample rates up to 48kHz. The interfacing also works with mobile devices as long as they have USB ports or a suitable USB adaptor. USB channels 1+2 carry the output of the amp with all processing, while USB 3 and/or 4 carry the dry guitar sound with no processing — in other words, a DI feed which may be used for reamping later. There's also a phones output (which mutes the speaker when phones are plugged in) and an aux input on a TRS mini-jack for play-along music.

Standard 5-pin MIDI is available only on the larger two models, though MIDI over USB is available for all three and facilitates remote preset changing plus controller assignments for virtually all parameters. The TRS footswitch jack, for use with an optional Line 6 LFS2 footswitch, is also located on the rear panel, as are the aux in jack, the phones outlet and the effects loop's send and return jacks. A Mode switch determines the function of the effects loop return, allowing for either a conventional effects loop setup or for the return jack to be used as a direct input to the power amp stage, for use with an external amp emulation pedal such as a Line 6 Helix. A separate footswitch jack input dedicated to the Boost would have been welcome, though you can reassign one of the existing footswitch functions to Boost via the free Editor app if required.

The effects section offers a choice of six reverbs, any one of which can be used at the same time as one of six delays, six modulations or six pitch effects.

The reverb amount is controlled by the reverb knob, with a button for entering selection mode, after which the amp model selector switch is used to choose the reverb type. The remaining effects are selected and controlled in a similar way using the FX knob, and a couple of buttons to allow for tap-tempo effect adjustment. For live use, the aforementioned LFS2 footswitch offers channel changing and effects on/off switching, but its function may be reassigned using the Catalyst Editor. There's also a noise gate, activated by pressing and holding the Boost button and turning the Boost knob.

While the amp model types don't seek to directly emulate specific amplifiers, the Chime setting does hint at a Vox AC30 and the six options cover pretty much all styles, with a nicely judged touch responsiveness that makes you want to keep on playing. The effect types also offer a generous range of impressively high-quality treatments, many emulating specific pieces of hardware, for example an Echoplex Echo, a Uni-Vibe, an MXR Phase 90, a Tycobrahe Octavia, a Leslie rotary speaker and so on, while the reverbs go from spring to shimmer. For deeper editing, the Catalyst Editor allows the virtual mic and speaker setups to be changed as well as providing the option of

adding amp hum and varying the power supply sag. It also allows for editing the amp and effects settings, as well as saving any of your presets to the amp or offloading them to a computer, but again it avoids getting over-complicated.

The bottom line is that despite its low price, this amp is very user-friendly, it sounds musically satisfying and its generous effects section could save you having to invest in a pedalboard. Its USB recording option and low-power mode also make it a perfect fit for home studio use. *Paul White*

$ $299.99.
W https://line6.com

Walrus Audio Fable

Granular Delay Pedal

I'm a real fan of granular delays, as they can transform the most innocent of sounds into a complex and musically engaging texture. If granular effects have a downside, it's that they can end up being quite complicated, but in the Fable pedal, US-based Walrus Audio have come up with a pedal whose five switchable preset program effect types interact with the other controls to provide a range of effects that range from smooth and hypnotic to glitchy and agitated, without becoming excessively chaotic.

The Fable is a mono-in, mono-out pedal that can accommodate guitars or keyboards. It requires a 9V DC, centre-negative, 300mA or greater power supply, ideally from a good quality regulated pedal power supply unit. All the connections are on the rear panel rather than the sides so there's no wasted pedalboard space. A 'hold down a switch, power on' sequence can be used to switch from trails to no trails when bypassing the effect.

Internally, the pedal uses two separate DSP engines, each with its own analogue feedback path, and in addition to the expected bypass footswitch, there's a tap tempo switch with flashing tempo LED — this or the Time knob can be used to set the delay time. This dual-engine approach makes it possible to combine fairly conventional tap-tempo delays with granular textures, though the function of each engine changes depending on the algorithm currently active.

There's no system for storing programs and no USB connection for updating firmware or storing patches to a computer, but the way the preset programs have been chosen makes it very easy to quickly dial in the desired sound. Indeed, if you set all the controls to halfway and then simply run through the five program switch positions, you'll find you still have a usable setting for each. This is particularly important for live performance as having too many parameters, or parameters with too wide a range of adjustment, makes it difficult to get predictable results.

Some of the control functions do change sightly when different programs are selected so a quick read through of the manual is recommended. But generally, Feedback sets the gain in the feedback path of the first DSP stage, while Regen does the same for the second DSP. Modulation adjusts the depth of pitch modulation, but if you want to change the modulation rate, you need to hold down Bypass while turning the Mod knob. Mix controls the wet/dry balance. On the second row, X generally controls the grain size. Time sets the delay time but also affects the granular buffer size (other than in program IV where it controls time-stretching of the grains). Tone places variable frequency filtering in the effect output and the feedback paths in the form of a low-pass filter with slight resonance. At higher settings a gentle high-pass filter is added, making the repeats sound brighter and thinner. There's also a momentary pitch Dive/Rise effect that is triggered if the Tap switch is held down. The direction of the pitch dive can be changed by pressing Bypass and Tap at the same time.

Program I sets up a reverse delay feeding into a reverse granular engine, so reversal and re-reversal occur in the feedback paths to add an organic complexity at longer grain sizes or a more pronounced stutter at shorter grain sizes. The result is something like a tape echo machine, albeit with a tape that the dog has chewed! Program II uses a forward delay fed into an octave-up granular engine achieved by playing back the grains at double speed. These build up in the feedback path creating a shimmery, pulsating delay that becomes a little more disruptive and chirpy at shorter grain sizes.

Program III is similar except the grains are shifted down by an octave, creating a surprisingly warm and involving sound with the low end pulsing in and out of focus. Program IV runs two multi-tap granular algorithms in series, where Time controls time-stretching of both granular algorithms to produce dense, reverb-like effects underpinned by granular flutters. At the maximum grain size this can produce a very smooth reverb-meets-beaten-up-delay effect. That leaves program V, which

feeds a delay into a granular engine that produces random octave up, octave down and randomly reversed grains. Here, X sets the rate of randomisation. The result is less chaotic than it might sound and again adds a very musical, pulsing texture to the sound, almost like the sound a tape makes when being fast wound while still in contact with the play head.

I should also mention the Mod control, as adding around 30 percent modulation adds warmth and movement to all the programs. In most cases setting the Mix control at between one quarter and one third up produces the most musical results, but there are no rules here — if it sounds good then it is good.

While most pedals tend to target guitar players, granular processing of this type can add a lot of character to basic synth sounds too. I also found that the Fable worked particularly well with Ebow guitar for ambient improvisations. It could be argued that some granular pedals, and certainly many granular plug-ins, provide a greater range of treatments but I think that Walrus Audio have done the right thing here by providing a very usable set of preset programs that can be adjusted over a fairly wide range but without allowing the user to get into too much trouble. Excessive granular processing can easily produce a disjointed, glitchy wall of noise but I suspect that isn't what most users would want from a pedal that is designed to be controllable in a live environment. The more I play with the Fable the more I like it, and it would slot beautifully into my live performances and ambient soundscapes. *Paul White*

$ $299.
W www.walrusaudio.com.

Arturia KeyLab Essential Mk3

Controller Keyboard

Arturia's new KeyLab Essential focuses firmly on computer control, with ambitious DAW integration.

ROBIN VINCENT

The KeyLab Essential Mk3 is the latest in Arturia's budget-friendly range of controller keyboards. The most significant new feature is the deeper integration with a range of DAWs, which I'll focus on for this review. But first, let's give its standalone facilities a once-over.

The 49-keyed KE3, which I'm reviewing here (there's also a 61-note version), has a good weight to it and is the perfect size for me. It's pretty solid, although entirely plastic. The chassis does flex a bit under chord stabs or when you are testing the aftertouch — before you realise it doesn't have any. The off-white (it's also available in black) is nicely sedate, juxtaposed with the shiny-as-heck new mid-panel display and posh-looking data encoder. The controls' flow and order is a nice refinement over the Mk2, and those lazily curved corners make it feel fresh and stylish. The feel of the keys is good, and they have a decent synth-style resistance. It's compact enough to sit between my keyboard and monitor without knocking everything off my desk, and while there's no wasted space, it isn't overloaded with controls.

The KeyLab bucks the modular-friendly trend with only a single MIDI output port and pedal input looking very lonely on the back. Computer connection is USB-C, and I appreciate the angled cable, which means I can butt the back up against my monitor stand without the cables getting in the way. Sadly, though, Arturia have put the other ports to the centre, defeating the good idea of the angled cable.

Analog Lab V

The ubiquitous Ableton Live Lite is bundled with the controller, along with a piano from UVI and one from Native Instruments. There are also some music lessons from Melodics and samples from Loopcloud. But the essential item is Analog Lab V. It doesn't look particularly stimulating but inside it's a fabulous feast of over 7000 sounds harvested from 24 vintage instruments. It's the best of the V Collection pulled into one place with an easy and immediate interface. And the integration with the KE3 is sublime.

You can navigate the whole laboratory from the KeyLab front panel. The encoder browses the library and selects sounds, while buttons let you jump between categories and styles. The knobs and faders hook into two rows of pertinent parameters, and you have a decent hardware synth experience.

Other features are common to most MIDI controllers these days, like a Chord mode, Scale options and Arpeggiator. These are all focused in the display via a few long button presses and a bit too much messing around with the encoder.

The KE3 has three primary programs or modes: Arturia mode for Analog Lab and

Arturia
KeyLab Essential Mk3
From $219

PROS
- Good looks, smooth layout.
- Seamless Analog Lab V integration.
- User-configurable MIDI maps.
- Solid DAW integration.
- Undo/Redo button.

CONS
- ARP/Chord/Scale options need a bit of menu diving.
- No way to shift control to other devices in Bitwig/Live.
- Cubase integration is a little limited.

SUMMARY
A clean and tidy refresh that brings decent DAW integration to an already deeply embedded Arturia control party.

other Arturia synths, DAW mode for DAW control, and User mode for regular MIDI. You can create custom MIDI maps for up to six devices in the highly configurable Arturia MIDI Control Centre.

DAW Integration

Arturia's scripted connections aim to give you a seamless, joyful, mouse-free DAW-controlling experience. They've damn near pulled it off, too. At the time of writing, the KE3 has a special relationship with Bitwig Studio, Cubase, Ableton Live, FL Studio and Logic Pro. Other DAWs are planned, but for now, the rest will have to lean into the good old MCU/HUI protocol. I checked out the integration in Bitwig, Cubase and Ableton Live, and here's how that went.

Bitwig Studio

Arturia assured me that the installation should be automatic and seamless, and it wasn't, but these things rarely are. However, once I installed the KE3 script, the integration was a very beautiful thing.

The KE3 controls the mixer faders and panning, scene firing and clip launching, browser navigation and device control. The transport control was also fully functional with play, record, stop, forward, reverse, tap tempo, metronome and loop on/off. The greatest button of all, though, was the Undo button, which will come in for some serious use.

The first button on the screen launches the Browser. You can use the data encoder to scroll through your various plug-ins and devices and use a pair of buttons on the display to shift categories. Bitwig has a very clunky sound preview facility that the KE3 had to wrestle with. The idea is that on the first click of the encoder, you get a preview of the selected synth, and then it loads on the second click. It just doesn't really work; the first click does nothing, and so you end up just having to click twice every time.

Once you're in a nice big project, then the KE3 really shows its quality. A single button on the display swaps the bank of faders and knobs between Mixer and Device modes. In Mixer mode, you get level and panning control and can move between banks of eight tracks. The ninth fader and knob always control the master output. You get little white triangles in the corner of the volume sliders to show you what's currently under control, which is a super-helpful feature that Ableton doesn't share.

Device mode flips the faders and knobs into controls for Bitwig's own synth and effect devices. You use the Part button to move between different pages of parameters and all the current controls are displayed along the bar beneath the device. It's really very good, and the way you can flip between controlling the mixer and then, say, the Polysynth is seamless. However, there doesn't seem to be any way to jump to another device. The controls will only affect the selected device, and so to move to another in an effects chain or back to the instrument after editing an effect, I have to drop to the mouse. The implementation is excellent; it's just not completely mouse-free.

Device mode isn't automatic for third-party instruments or plug-ins, not even Arturia's. It's a little strange because while in Device mode, you can map the controls to the parameters in the Bitwig Device version of the plug-in but not to the controls in the GUI. For that, you can swap to a User Program and use regular MIDI control. You can swap to Arturia mode for Analog Lab and other Arturia synths for a more seamless synth experience.

The last feature is scene and clip navigation, which is done through a combination of the encoder and the bank of pads. The pads can be used for playing notes and triggering drums, and they feel good for that, but they can also be a just-about-enough-pads-to-be-useful clip launcher. Bitwig draws a 4x2 rectangle in the clip launcher, and the pads light up with the right colours to reflect the active clips. You can then launch them with a tap or record into empty ones. The encoder knob moves the rectangle through the scenes, and you can launch a whole scene with a push. I got very confused on my first play because the pads were not launching the clips I expected. I then realised that the KE3 assumes you'll be in the vertical Ableton-style Mix Panel view, not the horizontal Clip Launcher view, where I spend most of my time. Once in »

The 49-note KE3 measures 790 x 240 x 70mm and weighs in at a fairly portable 2.76kg. The 61-note version, at 890 x 240 x 70mm and 3.08kg, adds only 100mm in width and an extra 320g in weight.

» the right place, it works perfectly as a little launchpad and navigator.

The only thing missing, I would say, is the ability to change views in Bitwig directly from the KE3. To be able to pull up the mixer, focus on device parameters, open instrument GUIs and so on would be very useful.

Cubase

The first time you open Cubase with the script installed is very impressive. You are treated to a virtual version of the KE3 laid out in the bottom half of the screen setup as a MIDI Remote device. We have Device mode and Mixer mode, but they are not as comprehensive as the Bitwig or Live versions.

For the selected track (and you can navigate between tracks using the encoder) the eight knobs become Quick Controls for that track. These are sometimes mapped a bit strangely for instrument tracks, although that may be down to what the plug-in designer makes available. It is very easy to reassign the Quick Controls. The faders control the send level for that track — that's all eight faders being used for sends for one track. It feels like they could have been put to better use. Arturia instruments can use the Arturia mode for a more complete control experience.

In Mixer mode, you have level and panning control over just the first eight tracks. There are no banks to switch to in order to reach any other tracks; you just get the eight. The last fader and knob always controls the level and pan of the selected track.

There are some additional controls. Clicking the encoder brings up the instrument's GUI (could do with that in Bitwig). You also have Arm, Mute and Solo buttons for the selected track. The pads can double up as project commands for things such as adding instrument tracks, duplicate, render, freeze, chord events, channel settings and sync settings. It would be great if you could also do things like add audio tracks, or open the mixer, editor or browser pages. I feel there would be other things I would do far more often that would sit there better. I wonder if it would ever be possible to customise these commands.

The Cubase control is solid enough and can be easily bent to your will with a few tweaks to the Quick Controls, but it feels a bit simplistic when compared to Bitwig and Live.

Ableton Live

The KE3 slides nicely into Ableton Live's seasoned control environment. It's very similar to Bitwig but with a sense of Ableton's calm level of authority. In clip launching mode, our pads give us a 2x4 launchpad displayed as a rectangle over our clip stacks. The encoder winds us up and down through the scenes and launches on a tap, whereas a pair of arrows takes us left and right through our project, with the selected track always displayed by name. A handy Record Enable button lets us get quickly into a track, and potential clips for recording are shown on the pads in red and are a tap away from capturing your musical thoughts. It is very smooth.

For device control, the knobs and faders are automatically assigned to the first device in the chain for that track. If you are using Ableton instruments or effects, then that works perfectly. For third-party instruments, they are in a similar predicament to Bitwig. The device controls won't work on the instrument or allow you to learn them. However, you can add a couple of parameters to the assignable X/Y pad, and you'll find the first two knobs are then mapped to those controls. If you opt for the User Program mode, then regular MIDI is returned, and you can MIDI Learn the KE3 controls to whatever parameters you wish. This was the same for Arturia synths except for Analog Lab V, which maintains its unfaltering integration when you switch to Arturia Mode.

Mixer mode is exactly as expected and has multiple pages for further banks of eight faders and eight pan pots. Pot and fader number nine always control the currently selected track.

As we found in Bitwig, the main missing piece is the inability to jump to the next device. You've often got chains of instruments and effects, and it would be brilliant to be able to navigate across to the one you want to control. The mouse is the solution, but if I'm having a lovely time playing in the KE3 control environment I'd prefer to stay there.

Conclusion

I really like the refreshed look and layout of the KeyLab Essential Mk3. It's well-proportioned, thoughtfully laid out and very tidy. I use the V Collection quite a lot, so I appreciate the consistently good mapping, and playing with Analog Lab V is thoroughly enjoyable. The DAW integration, particularly in Live and Bitwig, is excellent considering it's at the mercy of what the DAW can do. It feels as competent as what you'd find with Nektar keyboards but with the added integration of Arturia's own synths. If there were just a way to shift focus onto different devices in Bitwig/Live directly from the KE3 then they would have absolutely nailed it.

$ 49-key version $219, 61-key $269.
W www.arturia.com

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Zynaptiq Orange Vocoder IV

Vocoding & Effects Plug-in

This enduringly popular vocoding plug-in has been treated to much more than a graphical refresh!

PAUL WHITE

We reviewed the earliest incarnation of Orange Vocoder, then made by Prosoniq, way back in the year 2000 — so this has to be about as close to 'classic gear' as is possible for a plug-in to be! It's come a long way since then, though, and today, in the hands of developers Zynaptiq, it has been transformed into a seriously impressive sound creation tool. In fact, Orange Vocoder IV stretches way beyond the boundaries of conventional vocoding — it can be used to create sci-fi voices, ethereal vocal washes, drones and much more besides. As well as the vocoder section having been expanded considerably (there are now 24 different vocoding algorithms), there's a host of other toys, including pitch correction (now included as a separate processing module), a new formant shifter, an audio freeze feature and a comprehensive synthesizer section, tailored to the needs of vocoding. There are vocoding algorithms based on analogue modelling, independent component analysis (ICA), LPC filtering and wavelet transforms, plus there's the addition of unison and monophonic legato synth modes, numerous modulation options, and even an intelligent preset generator that uses 'smart randomisation'. A new filter bank release/reverb parameter called Traces also adds to the creative side of the processing.

Zynaptiq Orange Vocoder IV

$199

PROS

- Excellent range of vocoding effects and much more besides!
- Impressive pitch and freeze effects.
- Very intuitive.

CONS

- You'll need a decent CPU!

SUMMARY

Orange Vocoder IV is a special effects powerhouse that offers much more than most vocoders. Despite its depth, Zynaptiq have managed to make it all really easy to use.

All of this advanced processing does come at the cost of relatively heavy CPU use. On a modern machine you should be fine, but if you're in any doubt about the capability of your computer you can always try the free time-limited demo to see how well it fares (and it's worth remembering that increasing buffer size can reduce the strain at the expense

There are lots of vocoding algorithms, and they can be auditioned click-free, by hovering the cursor.

» of some additional latency). AU, AAX Native and VST 2.4 and 3 plug-in formats are supported and authorisation can be to a physical iLok dongle or a specific computer. Mac support is for OS 10.15.x or newer running on an Apple Silicon or an Intel CPU (two cores or more, i7 or better recommended), and for Windows it's version 10 or newer running on a CPU with at least two cores (again, Intel i7 or better recommended).

There are actually two versions of this plug-in; both are included in the purchase price, but this isn't the usual case of 'full fat' and a less feature-rich 'light' version. One is called, simply, Orange Vocoder IV, while the other is Orange Vocoder IV SC, the last two letters being an abbreviation of 'side-chain' and they cater both for the different MIDI and audio routing systems of various different host DAWs (for example, you need the SC version to route MIDI to the plug-in in Logic Pro) and for when you wish to use an external stereo signal as carrier for the vocoder, in place of the in-built synth. When you wish to use a mono signal as external carrier, the regular, non-SC variant of the plug-in can use its stereo input as a dual-mono configuration, with the left and right channels carrying the modulator and carrier signal.

The effects section includes modules for a freeze effect, pitch quantising, reverb and chorus.

Features & Operation

Following the lead of some other Zynaptiq plug-ins, the GUI hosts a series of separate tabbed pages and the first, called Overview, shows how everything's connected. Other tabs are called Synth, Vocoder and Mix & FX, which offer more detailed editing of the modules shown in Overview. A piano keyboard is shown across the bottom, with a selector at the left to switch between entering notes on the GUI keyboard or playing from a connected MIDI keyboard.

In Overview, the user can switch modules in or out of the signal path, and has direct access to a limited number of key parameters. It's also possible to solo or mute the various modules from here.

A Quick Setup button at the top opens a graphical menu, with a choice of ready-made Vocoder, Pitch Quantizer, Synth, Vocoder with Pitch Quantize, Synth plus Pitch Quantize or Dual Input Vocoder routings. A three-input mixer just prior to the Mix-FX block takes its sources from different places, depending on the selected routing; for example mixing the input voice with the synth sound and the vocoded sound or, in other modes, being fed from the pitch corrector or the Freeze module. A familiar dice button allows for random sub-presets to be loaded into the Synth, Vocoder and Freeze modules.

In each tab, you can click the name of a module box to navigate to its edit controls. Unused modules are greyed out and each module has access to its own sub-presets via a menu. When a sub-module is opened to show additional controls, an orange line appears around it when the cursor is hovered over it and, for example, in Synth view the bottom of the GUI changes from the keyboard view to show the additional controls.

The two-oscillator synth can run up to 64 voices, and offers a number of analogue-style waveforms as well as some vintage digital waveforms. A unison mode supports up to eight layers as well as Detune and Cluster modes, while Mono Legato with glide is also supported. The synth waveforms may then be processed via two types of ring-modulation, hard-sync, through-zero FM, and no fewer than seven types of polyphonic distortion. The synth also features a resonant low-pass filter stage with slopes of 6-24 dB/octave. Modulation options include both mono and polyphonic LFOs that can be cranked up into the audio range if required. There are also attack-decay envelopes and envelope followers, as well as more esoteric offerings such as zero-crossing trackers. Additionally there's a four-destination macro function to allow multiple parameters to be controlled simultaneously.

Most of us will be familiar with Auto-Tune-style automatic pitch correction, whereby incoming audio is forced to the closest scale note, but the processor included here has five operational modes. The less familiar Enforce and Polyforce modes use what's described as a wavetable-inspired approach, and they benefit from zero latency — importantly, this makes them practical to use in a live performance situation — and I achieved some lovely textural choral sounds using Polyforce mode, without even firing up the vocoder. Once active, the pitch correction is controlled by the incoming MIDI or the entered keyboard notes and includes any mono legato and glide settings.

The Freezer module, in the Mix & FX tab, comes with a range of preset

'frozen' sounds, but can also freeze the audio feeding the plug-in when you click the snowflake button. This locks the instantaneous timbre of the sound (until you unfreeze it) and, helpfully, there's the ability to save frozen timbres that you might want to use again. Also, as the Freezer comes before the Pitch-Q (quantiser) module, it's possible to play melodies based on the frozen sound. This frozen sound can be mixed with the synth or vocoder too, which often creates interesting textures that you wouldn't normally expect from a vocoder. While not offered as a routing option, mixing the freeze sound with the unprocessed main input is also possible, but only if the signal being fed into the aux input is the same as that fed into the main inputs.

In addition to its choice of algorithms, the vocoder section includes formant-shifting, expansion and stereo operation. Auditioning the various algorithms is straightforward, as you hear what each one sounds like while you move the cursor over its icon — you don't have to click to hear it. Once you find one you like, you can select it to load it. This 'hover and listen' approach is also used elsewhere, such as in the pitch quantiser, where it's handy for quickly auditioning the various modes. Simple controls adjust the formant as well as allowing variable amounts of high-frequency and unvoiced sound to be added to the output, which improves speech intelligibility. The unvoiced signal is actually synthesized, while HF Thru is a high-passed version of the modulator input. A small button in the lower left corner of the Vocoder box activates a compressor to help even out the levels.

The main purpose of the expander found inside the Vocoder module is that vocoded sounds can have a strongly reduced dynamic range — using the expander, it's possible to restore the original vocal dynamic contour, which can help the sound to sit better in a mix. This has a simple amount control and can be placed before or after the Traces section of the vocoder. The Traces section has controls for Time and Tone, increasing the release time of the vocoded sound to create a type of reverb-like, blurring effect. There's also a graphic-style EQ for shaping the sound.

Vocoder Verdict

Describing the 'sound' of this vocoder is really difficult, because it's capable of such a wide range of different effects. Obviously, any vocoder has first and foremost to nail the classic vocoding effects, and if that's what you want I have to say that the intelligibility you can get from Orange Vocoder IV is outstanding. But this is not just a vocoder: it is a powerful sound-design toolkit wrapped up in an intuitive user interface, and you can take it far beyond vocoding. When something more experimental is called for, you can set up voices that sound variously robotic, grainy, demonic, ghostly... and so on, while the freeze and pitch quantise sections add a new dimension, and can help create some wonderfully ambient textures, washy pads, drones, vocal harmonies and tuning effects. Being able to mix three sources at the output adds to the versatility, and if you need effects, you can add chorus and reverb to your sound too. Even the intelligent randomise feature is useful: it generally offers up usable results (which can't be said of all random dice buttons!). In short, Orange Vocoder IV has really impressed me.

$ Available separately ($159) or as part of the ZAP III bundle ($299). Existing users of any previous version of Orange Vocoder can upgrade to version IV for $99.

W www.zynaptiq.com

Modbap Trinity

Eurorack Module

The Trinity is a three-voice drum module with four separate synthesis algorithms to generate its sounds. Modbap promise a module capable of producing an astonishing range of drum and percussion sounds from right across the spectrum, and this promise the Trinity very much fulfils. With all three of its voices offering identical architecture, it can spread these sounds across the fundamentals of an electronic 'kit', focus on a limited palette with subtle variation, or just about anything in between; kicks, snares, blocks, hats, cymbals, shakers, bells, and more, it can not only do them all with ease but with surprising speed.

Each of the Trinity's digital voices begins with one of four algorithms: Block is a classically analogue-inspired synth voice with a sine and triangle core, named for its simplicity as a 'building block' of sound design. Next is Heap, an additive synth voice offering 12 detunable partials. Neon is an FM voice, and Arcade is a noise generator with more than a few distinguishing factors. The Trinity features controls for pitch and pitch envelope, as well as a state-variable, DJ-style low-pass/high-pass filter. There's a Clipper knob to add clipping distortion, and I was pleased to see Hold as well as Decay for flexible envelope control.

These controls ostensibly retain the same functions across the Trinity's different algorithms, save one or two variations when it comes to the Arcade noise generator (which also has the secret weapon of being able to ratchet). Things get more varied with the Grit, Shape and Character knobs, which endow their respective algorithms with extra oscillators, feedback, FM noise and all manner of other elements. Character sweeps through sine to triangle and up to a folding triangle in Block, detunes the 12 partials in Heap, increases FM depth in Neon and sweeps a band-pass filter through the frequency spectrum in Arcade, providing a quick, tactile point of focus for sound sculpting while maintaining a definite sense of 'no rules' openness. It's also possible to shake up the algorithms mid-beat, with 'random' and 'round robin' cycling options offering lateral variation for any or all of the Trinity's three voices.

Each voice has its own output, beneath which there is a single, mono Mix output. I confess this struck me as a slightly random inclusion on first impression. Switches toggle between sending each voice through its own output, the mix output, or both. I can imagine the design dilemma here: a stereo output might present a more useable mixer setup but wouldn't add much value, considering each of the three voices on the Trinity has its own output anyway. In practice, though, the Mix output's potential quickly became excitingly clear — and also helped to explain why Modbap included a HUE module with the Trinity unit they sent me.

The HUE, Modbap's 6HP mono effects processor, offers drive, a similar DJ-style low-pass/high-pass filter to that on the Trinity, cassette-flavoured tape saturation, a lo-fi bit-crusher and a one-knob compressor. It's a broad-strokes module capable of the kind of sounds that made Roland's SP-404 a hit among hip-hop producers, and while it's not a precision tool it sure gets decent results quickly. It's the perfect partner for the Trinity's Mix output, endowing its drum palette with deliciously bombastic parallel processing while its three individual outputs can be routed elsewhere for precise levelling. Be it compression, reverb, delay or just about anything else, the Mix output opens up a whole new territory for the Trinity's drum sounds, and the HUE is excellently demonstrative of this.

One of my favourite aspects of the Trinity is its aptitude for sloppy, sat-back, lazy-sounding beats. Its voices are stackable in twos or threes for surgical composite sound sculpting, but they can also be 'spread', which is to say they can be made to flam. This can be used to achieve anything from a dash of musical expressiveness to the kind of laid-back, barely-in-time snare shots that only the most audaciously confident beatmaker would attempt. The ratchets of the Arcade algorithm, too, can disrupt an otherwise locked-in beat with true vibe at the flick of a switch.

As far as standalone drum modules go, fair to say the Trinity is on the pricey side of things. I'd argue, though, that all things considered it delivers real value for money. It would be considered a high-punching module if it had but one possible algorithm, let alone four, and with Volt-per-octave tracking over its substantial synth engines can also hold its own in a melodic capacity. In fact, I can quite easily imagine furnishing a system with more than one Trinity, so characterful and flexible is its sound. *William Stokes*

$ $599.99

W www.modbap.com

Winter Plankton ZAPS

Eurorack Module

ZAPS has a sweet spot that sails like no other percussion module I've come across. Once you pass that threshold, you are off, and you'll have to be physically dragged away from the effortless pulsating slaps and giggles of this percussive funhouse. It's brilliant before you have any idea what you're doing or why it does what it does. In some ways, the depth and complexity that rides beneath the surface can detract from the enjoyment of playing with the thing, but after wallowing in the digital menus, you can always resurface and have a party on the front panel.

At the heart of ZAPS is a digitally controlled analogue synthesizer. It's designed for fast, percussive sounds that spit, splat, zip and zap, although it won't shy away from the odd melody. It has two analogue oscillators that push into each other, bringing out FM clangs and AM harmonics. All the usual waveforms are available for both VCOs. VCO 2, which takes the roles of modulator and noise generator, can have a second waveform for AM while the primary waveform is directed to FM. You have two simple Attack/Release envelopes with Hold, one routed to the modulation amount and the other to the VCA. Both envelopes can be roped into laser-zapping pitch-bending duties, from where our module probably gets its name.

ZAPS is all about percussion, so the focus is on triggering and changing the sound. It's linear and monophonic, so we've not got multiple channels of drums; instead, we have a stream of single events that can be worked into ever-changing patterns of sound. Some of this can be done through manual manipulation, but it really takes off when you swap in different presets for each trigger; that's where things get funky.

Preset sounds are held in banks of 12 slots. Each slot can contain a sound, so you load a bank and choose which sound you want triggered via the 12 buttons on the front panel. It's like having a monophonic drum kit at your disposal. The front panel then controls the selected sound. As with most digital front ends, the sliders don't necessarily represent the patch, but Winter Plankton can help with this: the sliders dim when they're not in the right position, so you can easily recreate the patch by moving them until they fully light up. That's quite neat; however, there are two pages of parameters, so nothing stays neat for long. From there the sound-design opportunities are deep and interesting, although I do wonder whether all the effort put into crafting the perfect zap is lost in the stream of triggers that inevitably gets pumped into it.

And pump in those triggers you will, because that's undoubtedly where the fun resides. With a single sound the front panel becomes a playground for tweaking and exploring. You can plumb in some assignable CV to keep everything moving, or you can set levels of randomisation to every parameter and have it find its own vibe. The next option is to play with the Morph control that takes all the parameters through every step from one sound to another. The third and funnest option is to CV-control the slots. Getting the right voltage to select the sounds can be a challenge, but I found a note sequencer works best for this as ZAPS can configure the slots to be selectable on each of the 12 notes of an octave. You can then program in a sequence of sound changes like a melody, giving you a perfect and very manipulable drum pattern. The front-panel control can become rather useless as it's constantly switching between sounds, but the chef's kiss of a feature is the Edit Group mode. With this engaged, everything you do on the front panel affects all the sounds, so even though they are very different you can still change the release on everything, mix the oscillators and mess with the modulations.

ZAPS is tremendous fun to use and can make sounds far beyond its percussive remit. The interface is playful and performative, but you can sometimes get lost in digital layers and button combos.

Robin Vincent

$ $589.99

W www.planktonelectronics.com

Modular Profile: Max Ravitz

Having joined the team at Moog Music back in 2015, product strategist Max Ravitz cut his teeth as both a musician and synthesist in NYC's independent synth boutique Control while playing transcendent electronica as Patricia. A key player in the development of the widely lauded Moog Mother-32, you may well have spotted Ravitz demonstrating its sonic capabilities as well as those of the Moog Matriarch and formidable Model 15 system.

On his entry into modular

I started dipping my toes into modular synthesis in 2012. I met this guy named Daren Ho, who was the co-owner of a modular synth shop that had just opened in New York, called Control. Soon after meeting him I started visiting Control and bought my first two modules, which were an original ALM Pamela's Workout and an original Make Noise Optomix. I became fast friends with the two owners of Control, and eventually started working there. I really owe a lot to those guys! They were incredibly supportive of my music career, and were amazingly flexible about accommodating my touring schedule, not to mention my gear collecting habits. I got to work with a lot of incredible musicians on staff there, plus it was great to be able to try so many modules. I have a lot of love for Control.

On his go-to modules

Interestingly enough, Pamela's Workout and Optomix were my first modules and neither have ever left »

» my case, although I've upgraded both to the current versions. Pamela's is never not useful, and the new Pro version just takes the flexibility to a whole new level. Optomix is my go-to low-pass gate, and I love the little tweaks they added from the original. For sequencing I use the Winter Modular Eloquencer constantly. I love Eloi who started Winter Modular, and that sequencer has cemented itself in my studio. For effects, the 4MS DLD [*Dual Looping Delay*] has lived in my case since it was released. The Lexicon PCM-42 is one of my favourite digital delays of all time, and it was designed by Gary Hall who also worked on the DLD. The guy really seems to know how to make a beautiful digital delay! Beyond those, I've really been digging the recent Bastl modules like the Pizza, Ikarie, Basil and Aikido. I love the Bastl vibe, their idiosyncratic approach.

On playing as Patricia

I started working on the Patricia project around 2008, but didn't get any traction until around 2013. I was living in New York at the time and did a random Craigslist trade with a local musician named Jan Woo. I traded him my Moog Prodigy for his Korg Polysix, and then sent him a song I made with the Polysix a few days later — an 'it went to a good home' kind of vibe. Jan liked the song, and invited me to perform at a venue he and a group of friends had opened called The Body Actualized Center for Cosmic Living — R.I.P. That was the first Patricia live show I ever played, and it was with a few artists on the label L.I.E.S. That was a huge turning point for me because I got to meet a lot of hardware producers working in a similar zone to me, and it allowed me to tap into a much larger pool of inspiration than I ever had before. I learned so much around that time, especially from musicians like Arp, Terekke, Jahiliyya Fields, Steve Summers and Bookworms. During that period my music found its way into Stephen Bishop's ears from Opal Tapes and he put out my first proper release, then everything developed from there.

On working with Moog

I started working with Moog around 2015 on the relaunch of their 5U modular systems. They initially hired me as an artist to perform in that launch video. Around that time they were also developing the Mother-32, and since I was a budding Eurorack nerd, they asked for feedback during the development process. I then ended up in the Mother-32 and Matriarch launch videos, and just kept working with Moog more and more as time passed. I even remember Mike Adams jokingly threatening to give me a job at one point because I was freelancing for them so much. Then around the end of 2019 that actually happened, and I moved from NYC to Asheville to start working for them, full-time. Now I'm the product strategist at the company, and I work with the product development team on instrument designs and broader product strategy.

On the culture of modular

This is a bit of a tricky one for me. I love modular synthesizers for their flexibility, and one of the things I would always tell customers at Control is that modular synthesis is amazing for building an instrument around your musical practice. When you buy a hard-wired instrument, the manufacturer has made all the decisions around how it works and sounds, and you have to adapt your musical practice to those decisions. You might also be paying for features that you don't like and never use, whereas modular synths avoid that pitfall because you can simply remove modules if you decide they're not what you want and rearrange your system to your heart's content.

This is a double-edged sword, however, because the ability to constantly reconfigure can distract from the ultimate goal of making music, and it can create the same type of decision paralysis that led me away from computer production. Of course, there are some people buying modular synths to explore sound and tinker, which I think is totally admirable and valid, but my personal goal is to make music. With that in mind I try to limit my modular case, which I've gotten better about now that I don't work in modular synthesizer retail! I find modular culture can be too centred on the fetishisation of the latest and greatest, and that can really reinforce the feeling that your case will never reach its final form. Lately I've been trying to just focus on creatively using the modules I have for what I need to do, rather than the untapped potential of what I could do with something else. I'm still occasionally lured into buying new modules, though, because most of the innovation in synthesis is coming out of the modular world, and I always want to try new approaches when it makes sense for my music. *William Stokes*

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CONTROLLING YOUR DRUM MIX

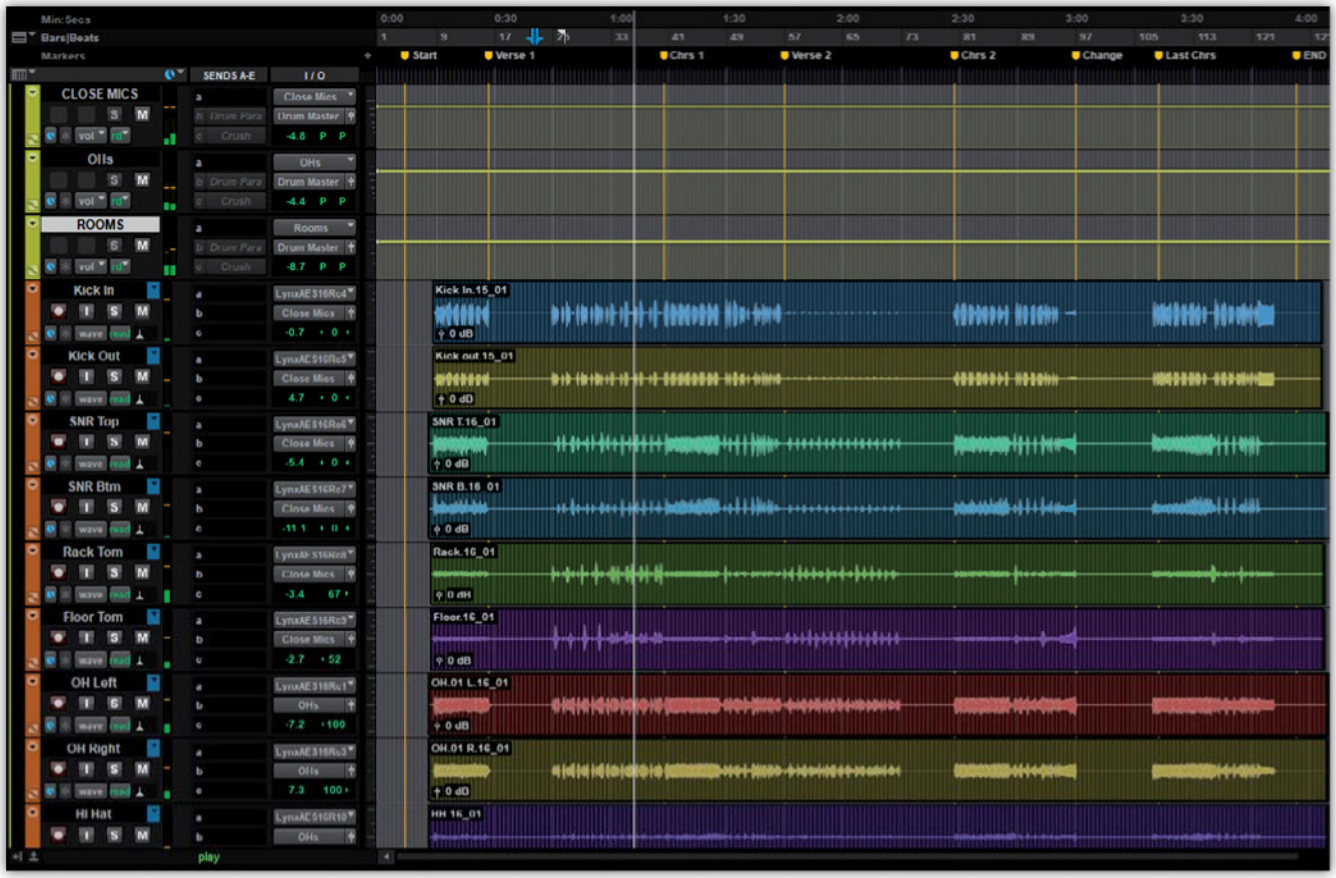
NEIL ROGERS

Drum mics can occupy a lot of channels in your DAW, and they can quickly multiply as you experiment with different processing — so it pays to impose some discipline!

Our DAWs offer us almost boundless creative freedom when mixing, but along with the infinite possibilities there's a downside: with the number of channels and routing options on offer, you have to be really disciplined if you're to retain control over your mix throughout a project. A recent project I worked on is a great example of where just such a lack of control presented a problem. Tasked with helping a client elevate the quality of a mix they'd worked on at home, I found myself unable to perform the most basic mixing move: I just wanted to turn down the snare! This was tricky, because the DAW session I'd been given featured an elaborate setup for the drums, with the snare routed to multiple auxiliary channels and various busses for processing. When I moved the main snare fader down it made little difference to the loudness of the snare. In fact, no fader did what I wanted. Hoping to avoid completely unpicking and rebuilding this drum mix, I brought up the level of the rest of the kit (thankfully, this was easy!) and then lowered the main drum bus fader. As a solution it worked, but it was inefficient and shouldn't really have been necessary.

Now, there's nothing wrong with creating sophisticated routing and parallel processing setups if they help you to shape the sound in a certain way, but it's essential that as you throw more and more ideas at your project, you retain control over all the key components of the drum sound. In particular, you need the ability to turn the key pieces of the kit up or down with a minimum of fuss — at any stage in the project and no matter how many channels you're working with!

With all that in mind, I want to take you through how I typically manage the routing for multitrack drums in my DAW. My setup is pretty simple in theory (there are no real 'tricks' here) but, importantly, I've thought it through and it's proved its worth on many projects. It helps me to keep my focus on what really matters: the big picture, by which I mean what the drums are contributing to the song. It also gives me the control I need to troubleshoot problems very quickly at any stage of the mix. Or,



The individual tracks of the drum kit are routed into three broad subgroups.

indeed, after I've finished it; for example if a client or mastering engineer comes back to request tweaks, I can go back into the project and make changes without having to get into a mixing mindset all over again. Yet it doesn't stop me experimenting with different techniques for creative effect.

Rule Of Three

Even at the very beginning of a mix, before any processing has been applied to the drums and I'm working to build a quick initial balance, I find it useful to have broad control over the sound of the whole kit using just a few group busses. That way I can mix fast and, as the mix progresses, if anything doesn't sound good/right (maybe I can hear a harshness in the cymbals or notice what could possibly be a phase issue), I can mute or solo these broad groups to help me locate the problem quickly.

I start by assigning everything to one of three stereo group busses, each of which is routed to a master drum bus, so that each 'family' of mics can be controlled and processed as a whole. Any panning is performed on the individual tracks. The busses are as follows:

Close Mics: Close kick, snare and tom mics, and so forth — any tracks that provide a focused capture of an individual, non-cymbal element of the drum kit.

Overheads: I route the main overheads and any cymbal spot mics to this group. I tend to rely on overhead mics to retain the 'lifelike' quality of acoustic drums, and this bus will typically end up being the least heavily processed.

Rooms/Effects: To this group, I route room or ambience mics, as well as any quirky 'effect' mics. It can be tricky to gauge how influential this family of mics will end up being in a given mix, and in the early stages of a mix I find it helpful to be able to quickly adjust their overall level. Later in the mix, I often find this bus useful for troubleshooting (eg. when a particular cymbal appears too loud during a certain song section).

Close Calls

As I get the mix up and running and start to develop a clearer idea of how I want to work on shaping the sound of the kit, I'll also create subgroups for the multi-miked kit pieces. (Yes, you could include these in a project template if you wanted to, but I generally prefer to keep my template simple, with just my main groups, and

Working With Samples?

Mixing drums can often be intimidating for newbie engineers, especially for genres and productions that generally require them to have a 'larger than life' quality, and it can be seductively easy to just reach for samples to get a gratifying drum sound. That can work. But I'd really like to encourage you to try and learn to get the most out of the 'real' drum recordings first, and if I've explained myself clearly here this setup will allow you to creatively feel your way around a group of microphone options — and can often produce surprisingly good results. If you do end up augmenting the drums with samples (or whatever funky processing your favourite YouTube guru is demonstrating!) then incorporating them in this framework will help you to get them working much more seamlessly with the rest of the kit, as well as making those finer balance moves towards the end of your mix much easier to perform.

The close mics are sorted into individual kit-piece groups for more focused mix processing.

then to flesh things out as I mix if the need arises.) Typically, this involves routing the multiple kick and snare mics to dedicated kick and snare groups, respectively, and I'll put tom mics into their own group too. These subgroups are all routed to the main Close Mics group I described earlier.

This setup effectively gives each instrument (not just each mic) within the kit its own fader, and with this all set up I might start processing the close mics, perhaps gating the close kick and snare mics at the subgroup level, and using EQ to generally shape the sound; the actual decisions obviously depend on the style of the production. It's not that I won't process the mic channels such as the kick in and out mics individually, but usually I'll have set the mics up at the recording stage to work together well, and too much processing can interfere with that relationship. Also I find it helpful to be able to shape the drum sound starting at the highest level possible (the drum bus), and then 'drilling down' when I need to get into more detail.

Parallel Lines

What I've discussed above is about getting the basic sound together. But getting drums to work as you want for any given song typically also involves a period of creative experimentation before you find an angle or specific tool that brings the particular drum sound to life, or imparts the distinctive vibe you're chasing. For this more experimental phase, I set up a few dedicated auxiliary channels for parallel processes (compression, saturation/distortion etc.) that can be blended in to taste to add excitement or interest. These tracks will all be routed directly to the main drum bus.

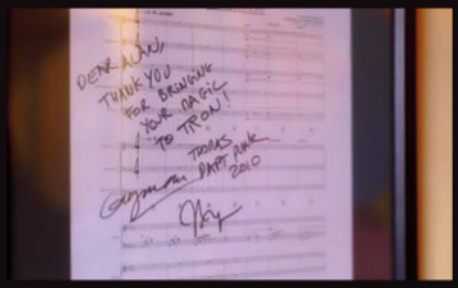
Although explosive parallel compression of the main drum bus (compressing the peaks very hard and blending this 'wet' sound in to taste with the uncompressed sound) can inject excitement into the drum sound, it often has unwanted side-effects. For instance, it can make the cymbals sound too harsh, or overly 'mucked around with'. To some extent you can EQ the compressor channel to mitigate that, but my routing setup allows me to use sends from the broad groups (Close Mic, Overheads and Room/Effects) to feed a different balance into the parallel compressor. Again, it's about quick and effective control.

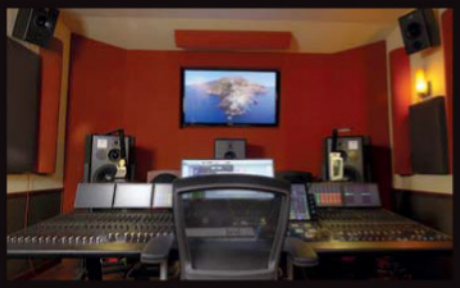
I don't tend to use much parallel processing on the individual kit pieces, and I think that's partly because by the time I've done what I want to shape the sound at the drum busses, main subgroup and these 'vibe' parallel tracks, there's often very little need. But, importantly, if I wanted to add a track for, say, parallel saturation of the kick close mics, it would be very easy for me to incorporate that within this overall routing framework — it would be routed to the group bus for the relevant kit piece.

As the mix develops, aux channels, routed directly to the main stereo drum bus, can help you get creative with parallel processing, without having to do too much processing at the individual track level.

Let It Flow

What should you take away from all this? You don't have to copy my approach, but I'd encourage you to think about the big idea — that you think through how you can retain the ability to quickly and easily shape and control the drums as a whole in the context of any song. I recently heard someone describe audio signal flow as being like tributaries, with various sources combining before they eventually all feed into the same river. That analogy works well here: every component in my final drum sound, including send effects like reverb, ends up flowing through one stereo drum bus; if you mute that drum bus, no drums reach the stereo bus. If there's just a little tonal issue bugging you about the drums (maybe they need a touch more presence, or a bit less low end) you can usually attend to that at the drum bus, and in a busy mix, if you decide you don't want the drums (or room mics, or snare...) sitting at the same level throughout, you can change that quickly and easily. But you can also work your way 'upstream' to get into as much detail as you want — without losing control.





SOUND ON SOUND

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Alan Meyerson: MIXING THE MOVIES

Pop producer and engineer Alan Meyerson was down and out when a chance meeting with Hans Zimmer took his career in a new direction. Thirty years and hundreds of film soundtracks later, Meyerson is Hollywood's first-call score mixer.

We visited Alan in his state-of-the-art Atmos mix room at Zimmer's Remote Control facility to hear his extraordinary life story and find out what's involved in mixing a blockbuster movie score. Then, in our special bonus feature, Alan takes us on a deep dive into his mix of Mark Mothersbaugh's soundtrack to the Marvel classic *Thor: Ragnarok*.

www.youtube.com/soundonsoundvideo

GROUND CONTROL

Understanding & Solving Ground Loops

Suffering from mysterious hums and buzzes in your studio? This down-to-earth explanation will help you identify and solve the problems — or hopefully avoid them in the first place!

HUGH ROBJOHNS

Some of the most frequent calls for help on the *Sound On Sound* Forum concern unwanted hums or buzzes emanating from studio monitors, usually with a statement that "it's something to do with grounding." That assumption is usually right, too — it often *is* something to do with grounding! But there are many ways that incorrect grounding/earthing can cause problems, and different fixes are required for different issues. It's also possible, of course, for there to be multiple different grounding issues within the same system at the same time. Figuring out the problem(s) requires some understanding of what grounding/earthing is supposed to do, as well as how different equipment is designed to be earthed (or not). But with the right approach, it's usually straightforward to work out the most effective way of interconnecting different kinds of equipment correctly and, even more importantly, safely. That last point really cannot be overlooked — people have died trying to fix grounding problems in some really stupid ways!

Solving grounding issues only requires a little basic fact-finding about the equipment involved, a plan of what's connected to what, some logical thinking and, it must be said, plenty of persistence and patience. It can often be a confusing, frustrating and iterative process, but I guarantee that following the concepts and techniques I explain below will eventually get you there.

Most unwanted noises in an audio system are caused by some form of external electrical interference getting somewhere it shouldn't. Sometimes that interference gets in via the mains cable, but often it's picked up as RFI (Radio Frequency Interference), so let's deal with that problem first.

The Faraday Cage

The best preventative solution for RFI is to protect sensitive electronics within a conductive box. This is called a Faraday Cage (or Faraday Screen) and derives from the work of Michael Faraday in 1836. In essence, it's nothing more than a conductive (metal or metalised) enclosure, which naturally stops varying electromagnetic fields (like radio transmissions) from passing in or out.

Audio equipment, like effects processors, mixers, interfaces and microphones, usually has a metal case that serves as a Faraday Cage, protecting the audio circuitry inside from external interference. A Faraday Cage doesn't have to be solid metal, either;

a metal mesh works too, provided the holes are smaller than the wavelength of the highest radio frequency to be excluded. Screening the electronics inside a unit is great, but the signals coming in and going out need protection from interference too. To that end most cables have a conductive outer shield of some type enveloping the signal wires, and that shield is normally connected to the equipment's metal enclosure when the plug is inserted. In this way the cable shield extends and links the Faraday Cages of devices when they are connected together. Although a Faraday Cage doesn't necessarily have to be earthed (grounded) — it isn't in the case of an aeroplane, for example — most are earthed, for reasons I'll explain shortly.

Putting this into a practical context, consider a balanced microphone connected to a battery-powered digital recorder. The metal case of the recorder forms a Faraday Cage protecting its electronics, and that Faraday Cage is extended by the shield of the mic cable to the microphone itself, where it is connected directly to the mic's case and pop-shield grille. External interference is therefore kept out of harm's way as the Faraday Cage protects and envelops all of the mic's and recorder's circuitry and associated wiring. As the recorder is battery-powered there is no direct connection to earth in this system (eg. via a mains cable) and so the setup is said to be 'floating' (in respect of earth). Nevertheless, it works perfectly without being earthed, as the audio signals pass from mic to the recorder entirely protected within a Faraday Cage and thus free of interference.

But what happens if the balanced microphone is replaced with an unbalanced electric guitar? The guitar's electrical components aren't normally protected inside a metal box — they're just crammed into a hollowed-out plank — and the signal voltage generated by the pickups isn't passed to the recorder via a pair of (balanced) wires safely protected within the cable's shield. Instead, the shield itself is employed as part of the (unbalanced) signal's circuit path.

Electrically, this arrangement looks very much like a TV aerial, so instead of the recorder's input circuitry looking for a signal voltage between two wires protected inside a Faraday Cage, it instead receives a signal voltage established between one signal wire and the Faraday Cage itself. Consequently, any interference picked up by the Faraday Cage is automatically added to the guitar's signal voltage (because they share the same circuit pathway). That's why unwanted hums and buzzes are a very common problem in this kind of scenario.

The cases around outboard gear and mesh grilles surrounding mic capsules act as Faraday Cages, which protect the signal and circuitry from RF interference.

What can be done to improve matters? Well, we could give the interference somewhere better to go than into the recorder's electronics — like a nice solid, clean 'earth'. The magic of the ground/earth is that it is a very large conductor that sits at zero Volts, and it is almost impossible to change that potential. So anything connected to earth will effectively be clamped at zero Volts. If the Faraday Cage is earthed, any interference it picks up will flow as current straight into the earth, and the recorder's electronics can now look for a signal voltage measured between the signal wire and the zero Volts of a clean earth.

Unfortunately, while earthing reduces interference collected by the Faraday Cage, the rest of the guitar's electrics are still a virtual TV aerial. But using copper foil (or conductive paint) to create a Faraday Cage around this circuitry inside the guitar's control cavity really helps to minimise the amount of interference getting into the control and switch wiring.

Groundless Laptop Studios

Studio DAW setups using laptop computers (or tablets) are typically also 'floating' systems lacking a solid earth. With balanced audio connections they can work perfectly well, but they often suffer in exactly the same way as the example above when connected to unbalanced equipment. Not surprisingly, then, providing a solid earth connection to the laptop's audio interface often removes troublesome interference problems completely — and it's easy to do.

The most convenient way of providing a good earth is simply to connect a device which has a protective earth connection via its mains cable. For example, most powered monitor speakers are Class I devices (see the 'Classy Power' box) with a protective earth, and that is normally extended back to the »

Terminology

The terms 'earthing' and 'grounding' are often used interchangeably, which can lead to confusion. In this article, I'm generally using 'ground' to refer to the electronic circuitry's 'zero Volt' reference and 'earth' when discussing a safety earth connection — the green and yellow striped wire in the mains cable which 'bonds' the equipment's metallic chassis to the incoming mains supply's 'protective earth' (PE). The protective earth connection is essential to the safety of some types of equipment (see the 'Classy Equipment' box) and ensures that, in the event of a serious internal fault, the mains power to that unit is disconnected promptly by a fuse or breaker. Regulations vary in different countries, but most require automatic 'breakers' to switch off the mains power supply if the current draw is excessive and/or if the currents on the Line and Neutral feeds are dissimilar — implying current is being diverted to ground somewhere (possibly via a human!). These automatic breakers can be fuses but are more often resettable mechanical switches known by initials such as RCBO, RCD, RCCB or GFCI.

» interface via the shields of the balanced audio cables used to connect them. So, just plugging powered monitors into your interface can provide the solid earthing needed to stop the system humming or buzzing. Connecting any other Class I audio device — a keyboard, an external signal processor, a guitar amp (they don't have to be switched on, just plugged in to the wall socket) — will work just as well. Having said that, having multiple earth connections can lead to another problem, as I'll explain shortly!

If suitably earthed external equipment isn't available, a dedicated earth connection can be provided instead. A safe way to do that is to use a dedicated earthing plug such as that made by Groundology, and I've described this arrangement in this article: www.soundonsound.com/reviews/groundology-earth-connection-plug. Groundology sell suitable earthed mains plugs for most parts of the world, but if looking for alternative suppliers don't be tempted to use the kind of earthing plug intended for static protection, since these have a built-in high-value resistor to restrict static discharge current flow and that's not appropriate for this application. The Groundology plugs provide a direct low-impedance connection to the protective earth terminal of the mains wall socket, and they can be wired to the sleeve terminal of a standard jack plug. Inserting that plug into any convenient socket on the interface provides a solid earth for the interface, allowing interference collected on the Faraday Cage to go direct to earth.

When Is Ground Not A Ground?

Given what I've said previously about the Earth's zero Volts potential, it seems logical to assume that all 'earths' or 'grounds' share that same 'zero voltage'. Unfortunately, the reality is that they don't... and therein lies the root cause of most ground-loop connection problems!

The reason all grounds are not all the same lies in the wires we use to connect them. All electrical conductors have some inherent self-resistance (and capacitance and inductance). Consequently, as soon as any current flows within the conductor, a voltage also develops along it, according to Ohm's Law: V=IR — the current (I) flowing in the cable, multiplied by the cable's internal resistance (R), results in a voltage (V) being generated along it.

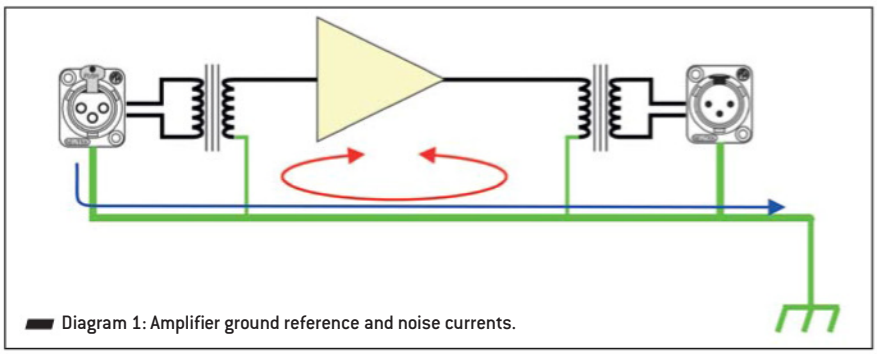

Diagram 1: Amplifier ground reference and noise currents.

Of course, the resistance within a cable is usually extremely small (milli-Ohms), but it still results in a small voltage being developed when a small current is flowing. This is significant because audio signals are, themselves, often quite small voltages (usually measured in milli-Volts). So, it doesn't take much current flowing in a sensitive ground path to generate audible hums and buzzes! What do I mean by a 'sensitive ground path'? Well, the electronic circuitry in most audio equipment works with unbalanced audio, where the signal voltage is measured relative to a 'reference ground' (see Diagram 1). Consequently, any noise currents passing along that reference ground will intermingle with the wanted audio currents, inherently adding unwanted noise to the audio signal. So, it's critical that noise currents are kept away from the audio reference ground. This is achieved mostly through good equipment design and situationally appropriate interconnection techniques.

Grounds: How Many Is Too Many?

The most common 'grounding problem', historically, is due to noise currents arising from 'ground loops' getting into equipment via the protective earths of their mains power connections. It's often simplistically described as having 'too many earths within a system', but there is nothing fundamentally wrong in having multiple earth connections if the audio equipment and its interconnections are designed and implemented correctly. Well-engineered studio installations all around the world confirm that fact. However, equipment isn't always designed or connected ideally, so we need to take pragmatic measures to deal with flawed systems.

Ground loops mostly involve Class I mains-powered equipment — that's equipment with a protective earth connection via the mains cable (see 'Classy Equipment' box). If the equipment is battery-powered, or Class II, or powered using a low-voltage external supply, it's unlikely to play any role in a suspected ground-loop problem. This is why it's important to identify the type of equipment being connected when trying to resolve 'grounding problems'.

So, what is this mysterious ground loop? Consider a simple bedroom studio with several mains power outlets on different walls around the room. The mains wiring inside the wall linking those sockets back to a Consumer Unit will be many tens of metres in length, and with such long cables there will inevitably be small but measurable resistances between the terminals of different wall sockets.

In a perfect world, there would be no current flowing along the protective earths of the in-wall mains wiring and, if there's no current flowing through the cable's resistance, the earth terminals in every wall socket would be at exactly the same (zero) voltage as the power station's local earth rod sunk into planet Earth. However, in the real world most mains-powered devices have mains filtering circuitry and/or a mains transformer, both of which introduce capacitance between the live, neutral and protective earth connections. Mains power is AC (Alternating Current) and so those small capacitances allow a small AC current to leak through to the protective earth. Consequently, mains-powered devices plugged in around the building all contribute small leakage currents into the protective earth wiring. This is entirely normal and perfectly safe, but the issue for us is that those small leakage currents create small voltages along the tens of metres of in-wall wiring, meaning that

the earth terminals of the mains outlets on different walls will acquire slightly different (non-zero) voltages.

So, we have various Class I mains-powered audio devices plugged into different sockets around the room, and that equipment is interconnected to pass audio signals. Remember, the audio cable shields are normally connected directly to the equipment's protective earth too, so just by plugging this equipment together and into the mains sockets we have created a 'ground loop' (see Diagram 2).

As we already know, then, there's likely to be a small voltage difference between the earth terminals of the different mains outlets because of leakage currents and cable resistance. The mains outlet with the higher voltage will force a small current to flow from that wall socket's earth terminal into the associated audio device (A) via its mains cable's protective earth. From there it travels along the (red) audio cable screen to the second audio device (B); and then via its mains cable's protective earth into its own local mains wall outlet. The current then flows back to the first wall socket via the building's in-wall wiring. Hence a classic ground loop — it's inevitable given the way we normally power and connect equipment in a studio.

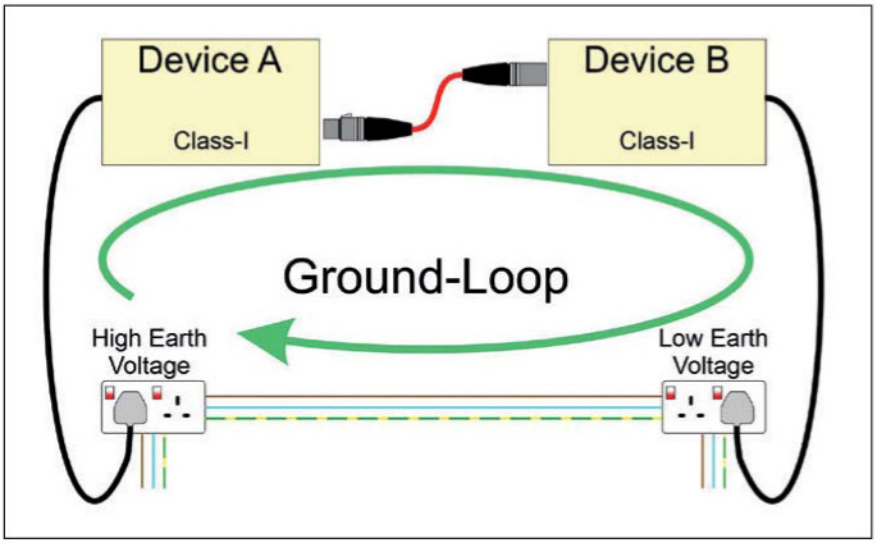

Diagram 2: A classic ground loop, formed when two directly connected devices are plugged into separate wall sockets on the same ring but with slightly different earth voltages.

Stars & Daisies

This circulating ground-loop current isn't a problem in well-designed equipment, as I've already stated (we'll come back to that shortly), but it still makes sense to minimise it where we can, and the logical first step is to remove the voltage difference between the equipment's mains connections, and we do that by minimising the area of the ground loop. For example, instead of plugging equipment into different mains wall sockets, it's far better to power them all from the same wall outlet, so that they share the same earth connection — this guarantees there will be no voltage difference to push a noise current around the loop in the first place.

Most studios have more than two mains-powered devices, of course, and multiple plug boards are typically needed to provide additional mains sockets. However, since everything plugged in is likely to leak a small current into the plug boards' protective earth wiring, small voltages can still develop, leading to different earth voltages in different plugboards. That situation is greatly exacerbated if the plug boards are connected in a daisy-chain, but minimised if they are connected in a star arrangement, thus greatly reducing the chances of ground-loop problems (see Diagrams 4 and 5).

Some do question the safety of powering an entire studio from a single socket, but here in the UK a standard wall outlet nominally provides 2990W of power (the available power depends on the supply voltage, so can theoretically range from 2808 to 3289 Watts in the UK). It's very unlikely that a typical project studio would consume anything like that much power, and it's safe to run an entire studio off a single wall outlet. For more peace of mind, simply add up the power ratings of every mains-powered item in the studio; the relevant figures can normally be found near the device's mains inlet, or from labels on wall-wart and line-lump PSUs. The most power-hungry devices will be active speakers or power amps, desktop computers, and large mixing consoles, each of which might require several hundred Watts. Everything else will probably consume tens of Watts at most.

Noise Currents Inside Equipment

I've said repeatedly that well-designed equipment using balanced audio connections isn't affected by circulating »

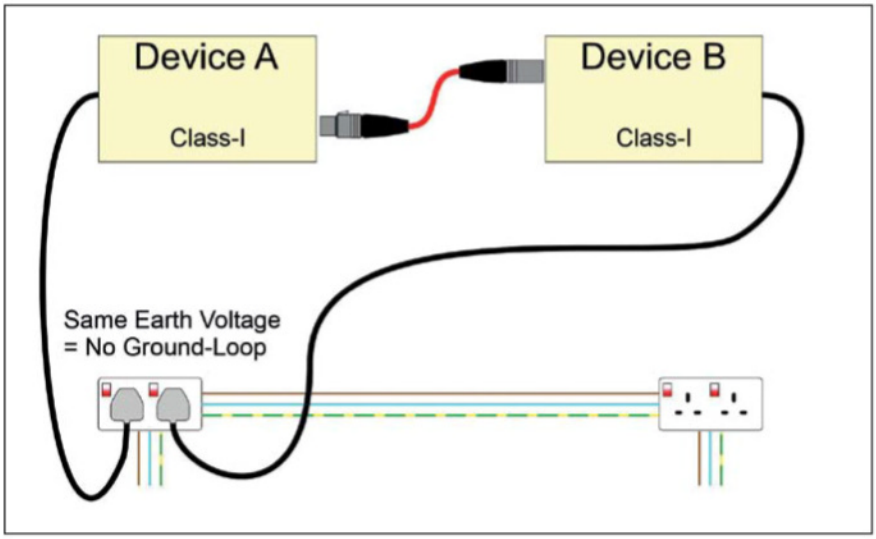

Diagram 3: The classic ground loop is cured by plugging the connected devices into the same wall socket — because there's then no difference in the earth voltage.

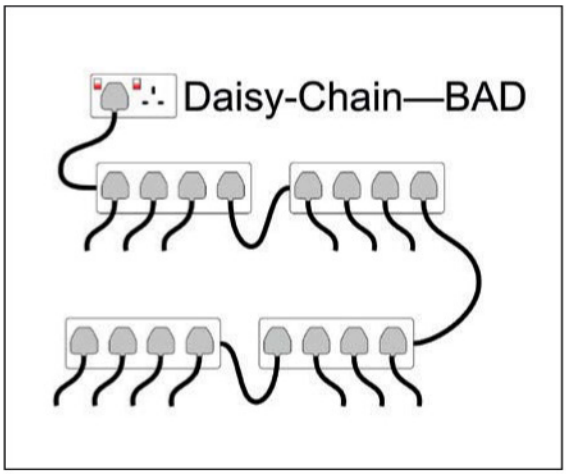

Diagram 4: Daisy-chaining distro boards is a bad idea for the same reason — at different stages in the chain, the earth voltage will be different, thus creating the conditions for a ground loop.

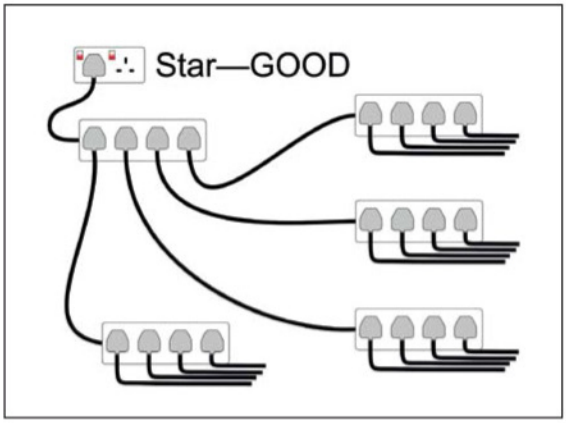

Diagram 5: Cascading distro boards in a star arrangement overcomes that problem — again, the sockets into which connected devices are plugged will have the same earth voltage.

» ground-loop currents, but in poorly designed equipment ground-loop currents can find their way into the audio reference ground to create audible hums and buzzes — hums because the noise current is derived from the 50 or 60 Hz mains supply, and buzzes because of the related harmonics that accrue on mains supplies.

Diagrams 6 and 7 illustrate part of a mixing console or audio interface. I've shown several transformer-balanced inputs feeding high-gain amplifiers (the yellow triangles). The amplifier outputs might feed output transformers (as here), or be mixed together, or be converted to digital — it doesn't matter for the purposes of this example. What does matter conceptually for this example, though, is that the input transformers convert between the balanced external connections and the unbalanced internal amplifier circuitry, which measures audio signal voltages against the reference audio ground (shown in green). The red loop represents the audio signal current path around the amplifier circuitry, while the blue line shows the external noise current path from the input connection going to earth. We've met this situation before...

As you can see in Diagram 6, the reference grounds of each input channel are connected to a 'star ground', which is itself earthed via the mains cable's protective earth connection. However, in this 'bad' design, the ground-loop or RFI currents carried on the input cable shields pass through the audio reference grounds of each channel's amplifier on their way to that star-ground point. Consequently, unwanted ground noise is inherently added to the wanted audio signal within each channel. A lot of traditional audio mixers were built in exactly this way, with pin 1 of the channel's XLR wired straight into the mic preamp's ground (simply because it was convenient). This has become known as the 'pin 1 problem' and much has been written about it in professional audio circles!

Avoiding this catastrophic problem is remarkably easy, as shown in Diagram 7: noise currents on the cable shields simply require a separate route to earth, bypassing the audio channel reference grounds. This is shown in the diagram below, in which the ground-loop and interference currents from the cable shields at the inputs are kept completely separate from the audio reference grounds, so can't get into the audio amplifier circuitry at all. This arrangement is typically achieved simply by bonding the shield terminals of the input (and output) connectors directly to the equipment's metal enclosure, maintaining the Faraday Cage concept discussed earlier.

So, by using balanced connections and constructing equipment sensibly, noisy ground currents are kept well away from the audio circuitry, and thus ground loops and external interference aren't problems at all. But what do we do with equipment that *isn't* well-designed, or when using unbalanced connections? In these situations, the best approach is to prevent the noise currents from entering the equipment in the first place, for example by breaking the ground loop.

Clearly, there are several places where a ground loop could be broken — but many of them are fundamentally unsafe! For example, disconnecting the protective earth at the mains wall sockets is obviously not acceptable, since it would defeat a primary safety provision. For the same reason, the protective earth cannot be disconnected in equipment power cables or their mains plugs — such outrageous 'solutions' have killed a lot of guitarists over the years!

There are many safe and simple alternatives and the easiest, often provided by the manufacturers of Class I equipment, is a 'ground-lift' switch. Class I equipment normally has a direct connection between the internal audio reference ground, the equipment's metal chassis, and the mains cable's protective earth — usually at a star-ground point. A ground-lift switch simply disconnects the audio reference ground from that star-ground, allowing it to 'float'. In this way noise current trying to get to the protective earth via the audio reference ground (as in the 'bad' equipment example above) no longer has anywhere to go — the ground loop is broken. Note, though, that the equipment's metalwork is still safely earthed via the protective earth connection in the mains cable.

If a ground-lift switch isn't available on a Class I device, the next easiest option is to interrupt the audio cable's shield connection in some way, and this approach also works if the noise currents are due to RFI. (This won't work for phantom-powered microphones, as the shield connection forms the 48V power supply's return path.)

The cable shield connection can be severed simply by opening the connector, cutting the screen connection inside the plug, and isolating it with insulating tape. This only needs to be performed at one end of the cable (usually the destination/input end) and it's an arrangement often referred to as 'one-end-only', which was a backbone of much professional studio installation wiring. If you do cut the shield connection, do remember to mark the

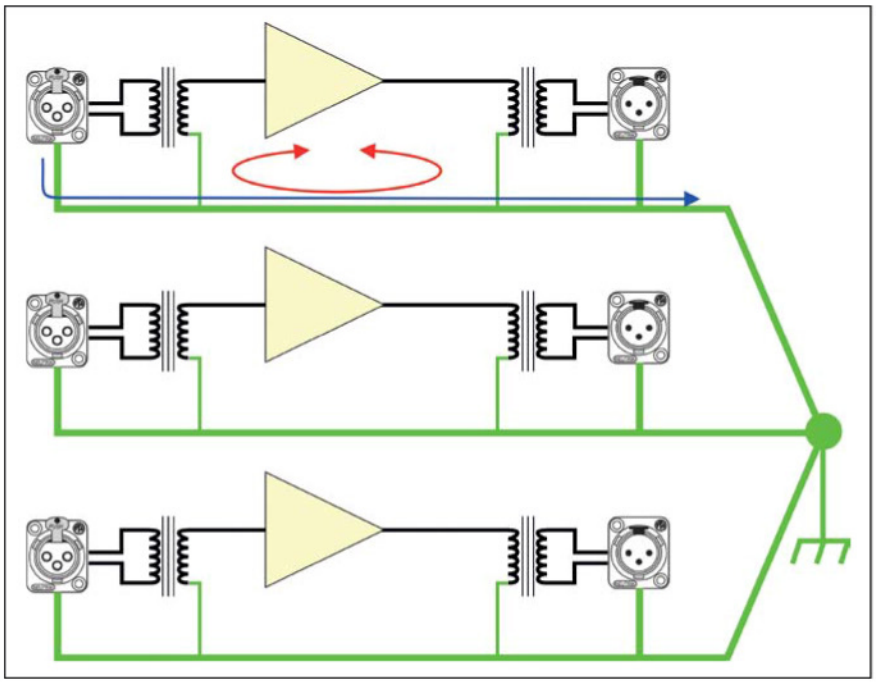

Diagram 6: Grounding inside equipment is also important. Here, despite the star-grounding arrangement between channels, a loop is created because the input cable shields pass through the audio reference grounds of each channel's amplifier on their way to that star-ground point.

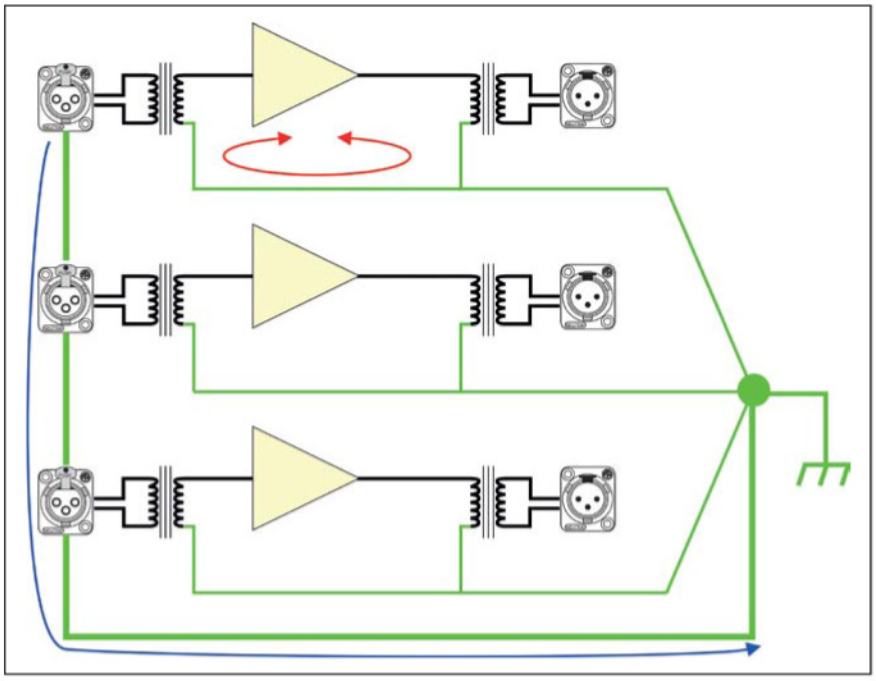

Diagram 7: Thankfully, curing the problem in Diagram 6 is trivially easy!

cable in some clear way, to avoid it being re-deployed later in a situation where the shield connection is required!

With balanced interconnections, the shield plays no role in transferring the audio signal, so breaking that connection doesn't affect the audio, just unwanted shield noise currents. The same cannot be said of unbalanced connections, though, and alternative solutions are required for solving ground loops with unbalanced equipment (see below). The one-end-only solution is also not applicable with battery-powered and Class II equipment, as the cable shield is used to share the signal ground reference.

If you'd rather not hack your cables, a more versatile though more expensive solution is to employ an audio isolation transformer box. The 1:1 transformer passes the audio signal as a varying magnetic field across the transformer's windings without any direct electrical connections (including the shield) between the box's input and output sockets. This breaks a ground loop, and prevents RFI currents from getting into the receiving device (Diagram 8). Audio isolation transformer boxes are available in various formats and prices from numerous manufacturers, and although ultra-low-distortion transformers are quite expensive, modestly priced options are often perfectly acceptable. For instance, I routinely use ART's DTI isolation box for identifying and resolving ground loops and related problems. It offers two channels with a plethora of connector formats, making it very convenient and versatile to integrate quickly into any system with balanced or unbalanced connections.

In general, transformer isolation boxes work best when installed at the destination end of a balanced cable run, but at the source end for unbalanced connections (where it can also convert the output to a balanced format, if helpful). Transformers are susceptible to external magnetic fields, of course, so do make sure that the isolation box is kept well away from mains cables and, especially, any mains transformers in nearby equipment — it's quite irritating if a ground-loop hum is replaced with magnetically induced hum!

When I suspect a ground loop or shield interference problem I usually employ a transformer isolation box to test my theory. If the unwanted noises go away when the box is inserted, I know they were due to noise current on the shield connections. If practical, I would usually then remove the isolation box and cut the original cable shield(s), marking them accordingly, because it's a much less expensive solution than permanently installing a transformer box, while also avoiding any possibility of a transformer degrading the sound quality. If the noises don't go away with the isolator box connected, I know the problem lies elsewhere!

Unbalanced Sources

The transformer isolation box is particularly helpful in resolving ground loops with unbalanced Class I sources (like some vintage

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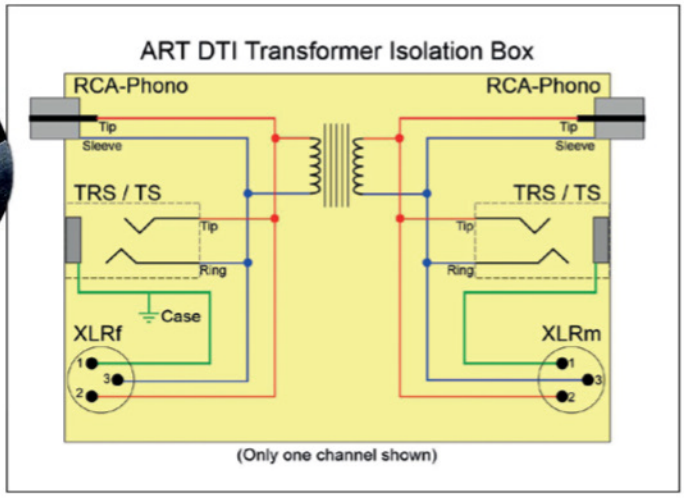


Diagram 8: even if not left permanently in the circuit, isolation transformer boxes such as the ART DTI can be very helpful in diagnosing or ruling out the possible causes of hum.

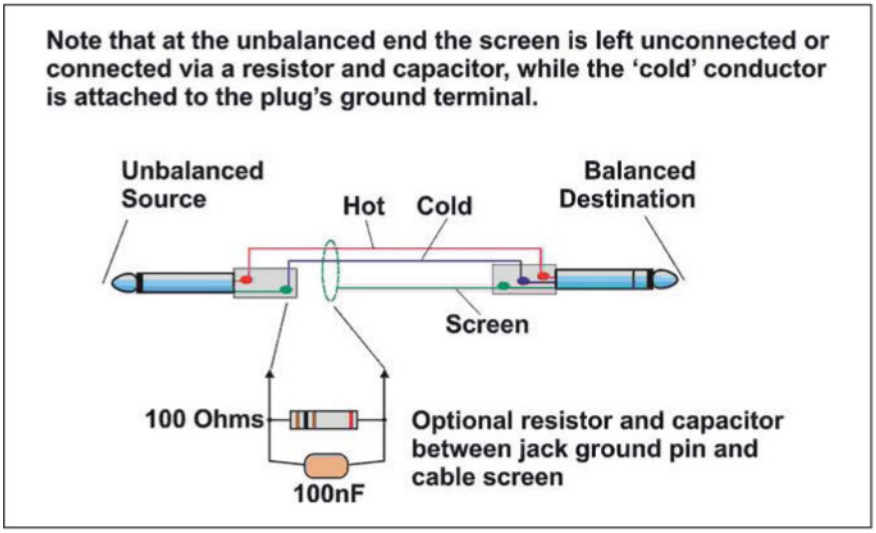
» synths). But if the unbalanced source is connected to a balanced input (like in a mixer or interface), another option is a 'pseudo-balanced' cable. Unfortunately, and despite their immense usefulness, I'm not aware of any commercial cables wired to this format, but bespoke cable-making services can construct them to order, and they're an easy DIY build for anyone familiar with a soldering iron.

A standard balanced cable (two-core plus shield) is used, with the balanced destination end wired as normal with a male XLR or TRS plug. At the unbalanced (source) end, the hot signal wire is connected to the unbalanced signal output (the tip of a jack plug, for example), while the cold signal wire is connected to the plug's shield (sleeve) terminal. In this way the source's signal voltage is passed directly to the balanced input, but there's no connection between the source's ground and the destination ground — hence no ground loop. The cable's shield is connected at the balanced destination end but left isolated (and insulated) at the unbalanced source end.

Many of my synth cables are built in exactly this way, and work perfectly well. However, the unterminated cable shield is effectively acting as an aerial to pick up local interference — if that's a problem, then better RFI protection can be achieved by connecting the cable shield at the source end's plug sleeve terminal via a small capacitor. The value isn't critical; anything under 100nF that fits inside the plug body will be fine.

Alternatively, a low-value resistor can be used to maintain some RF screening while reducing any ground-loop currents to inaudibility. 100Ω is a typical value in this application, but again it's not critical. If you have a large enough connector body, the resistor and capacitor can be used together in parallel.

It's important to note that a transformer isolation box and/or pseudo-balanced cable are only appropriate if the unbalanced source is a Class I device. If these options are employed with Class II devices, or those powered via wall-wart PSUs, they are more likely to create problems than to fix them! This is because such devices rely on receiving a ground reference directly from the destination device via the connecting cable's shield. Intentionally interrupting that ground connection leaves the source floating and acting as an aerial to collect and inject unwanted RFI.



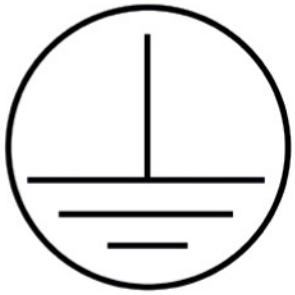
The correct wiring for a pseudo-balanced cable.

Plan It: Earth!

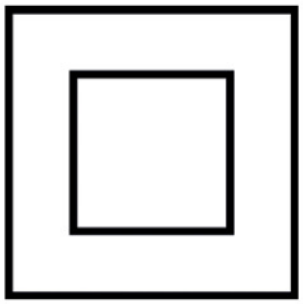
So, I've described some practical solutions for a variety of 'grounding problems' but how should they be deployed? I mentioned earlier the importance having a clear and detailed plan of how the entire system is connected together. I like to sketch it out on paper, with a note on each device identifying whether it's Class I (safety-earthed), Class II (double-insulated), or battery-powered and, ideally, where each unit is powered from (which wall socket or mains distribution board).

The plan should also show all audio connections between devices (multiple channels between two devices can be shown as a single thick connection for the purposes of ground loops), and specifying which connections are balanced or unbalanced. Include digital audio, digital clocking, USB, FW and TB connections, too. Although it is helpful to include ADAT, optical SPDIF and

Classy Equipment



The identifying symbol for a Class I device.



The symbol for a Class II device.

Most studio equipment is mains-powered, but there are different 'classes' of mains powering, which determine how grounding issues can be resolved.

Traditional devices with internal mains power supplies are built into a metal chassis connected to the protective earth via the mains cable. Products built this way are called Class I devices, and include most desktop computers, power amplifiers, large mixing consoles, professional tape recorders, vintage outboard equipment, and so on. They can be usually identified by an 'earthed' symbol on the identification plate near the mains inlet.

The electrical safety of a Class I product relies on two things: insulation around the live conductors, and exposed conductive parts being connected to a protective earth. If the insulation breaks down AND the live conductor makes contact with exposed metalwork, a very large current flows directly from the Line (Live) connection to the protective earth, instantly activating legally-required protection devices within the mains supply (fuse, RCBO, RCD, RCCB, GFCI etc).

Consequently, all Class I products have a three-terminal power connector — most commonly the IEC C14 inlet connector — and they MUST be connected with a three-core mains cable from a three-terminal wall outlet to maintain the protective earth connection. NEVER disconnect or circumvent the protective earth; it's extremely unsafe, quite unhelpful in resolving grounding issues, may be unlawful, and it could kill you in the event of a fault!

An increasingly common alternative arrangement is called, logically enough, Class II or 'double insulated', and is identified by a 'box-within-a-box' symbol.

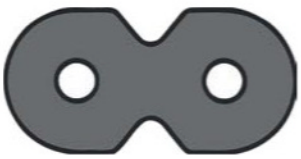
Class II equipment includes many musical keyboards, outboard processors, some active loudspeakers, hi-fi equipment, and much more besides. Electrical safety is ensured in a Class II product by using two independent forms of insulation around live components so that no single fault can result in the user becoming exposed to the Line voltage via exposed conductive parts. Consequently, Class II devices don't need a protective earth connection and therefore normally use a two-core cable and two-pin connector such as the IEC C8 (fig-8) or C18 style IEC inlets.

A subset of Class II is identified as Class IIFE, where 'FE' stands for 'Functional Earth'. These devices fully comply with the double-insulation requirements of Class II and are marked with the same box-within-a-box symbol. However, they have a three-core mains connection — most commonly using the IEC C6 (Mickey Mouse or cloverleaf) inlet.

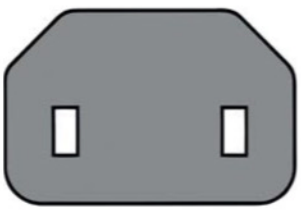
This Functional Earth is connected via a three-core cable to the wall outlet, as with Class I, but the Functional Earth's role is not to provide shock protection in fault conditions (which can't occur because of the double insulation). Instead, the Functional Earth helps mitigate electromagnetic noise and interference by providing a direct path to Earth from Faraday shielding. Class IIFE devices include some laptop power supplies and lots of computer-related and digital audio devices.

Class III devices are powered (or charged) via an external low-voltage mains adaptor ('wall-warts' and 'line lumps'), such as laptops, some effects units, keyboards and so on — anything with a low-voltage power inlet, basically. However, the salient point about these devices is that their mains power units and chargers are always Class II or Class IIFE, so we can consider them as being part of that category when resolving grounding issues.

An IEC C14 connector.



An IEC C8 connector.



An IEC C18 connector.

An IEC C15 'cloverleaf' connector.

MIDI connections as part of the system overview, these are all optically isolated and can't contribute to ground loops or RFI. All the others (bar a few specialist optical alternatives) involve screened cables that are normally grounded at both ends. With a good 'paper plan', you can see if the system as a whole is earthed, as well as anticipate likely ground loops and any unbalanced/balanced connection issues.

If fault-finding an existing system, I start by disconnecting absolutely everything apart from the computer, interface, powered monitors and monitor controller (if there is one). I then play some reference material to set a sensible listening volume and check there are no hums of buzzes. If this minimal rig is noisy and involves Class I devices I'd try an isolation transformer between the interface and monitors. If it's all Class II, I'd try an earthing plug.

Once this core system is quiet, I add the other equipment on the plan, piece by piece, checking again at every stage and correcting any problem introduced by that device using the techniques described above. With good equipment and balanced connections there probably won't be any problems at all, but by building the system up slowly, testing devices and connections one by one, it's usually straightforward to find and solve any issues. In contrast, diving into a complex, fully wired system that hums or buzzes can be a confusing and frustrating nightmare.

Ground-loop problems will always be worse with unbalanced Class I equipment, so try and plug the mains cables from those devices into the same plug board powering the computer/interface/mixer to minimise the loop area. Well-designed equipment with balanced connections can be plugged into more distant sockets as they won't mind circulating ground-loop currents!

Remember, nobody designs their equipment to hum or buzz, and there's usually a way of connecting them to be completely noise-free. It just talks patience, logical thinking and educated experimentation. Good luck!

Better MIDI Pianos

Editing Virtual Piano Performances

A few tweaks to your MIDI data can make software piano parts much more realistic — and much easier to mix.

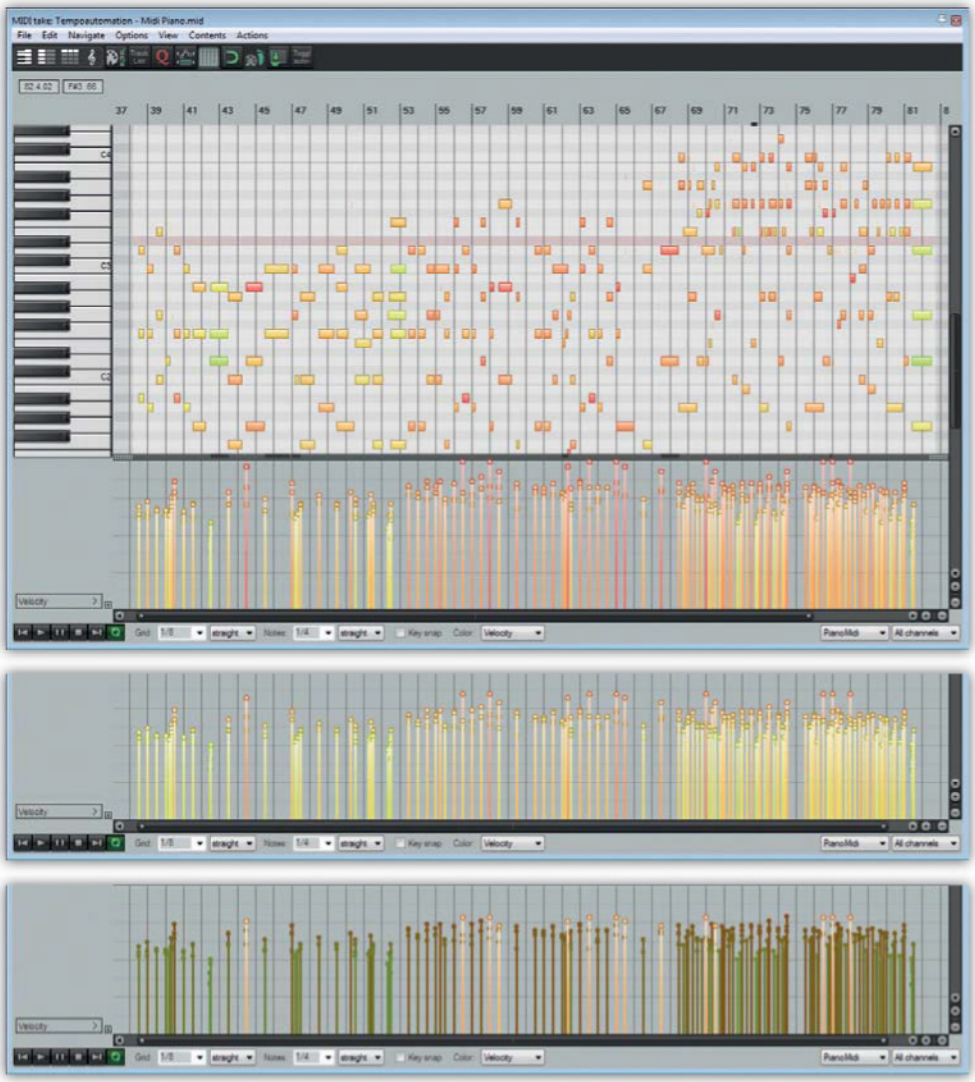
MIKE SENIOR

Recording an acoustic piano isn't a task to be taken on lightly, even if you happen to have access to a well-maintained instrument in a suitable venue, so it's little surprise that many project-studio users turn to MIDI-triggered software instruments instead. In my own mix work I encounter such simulations on a regular basis and, to be honest, the results often leave a lot to be desired in terms of realism and musicality. So in this article I'd like to recommend some tactics for improving your prospects when working this way.

MIDI Humanity

The first thing to say is that it's both extraordinarily difficult and extremely time-consuming to program anything but the simplest piano part by clicking the notes in with your mouse, so I'd encourage you to record your piano part as a real-time performance wherever possible. If your own performance or improvisation chops aren't up to the demands of the part, then I'd definitely consider enlisting a more expert ivory-tickler to help you generate the bulk of the MIDI data you need. Failing that, however, there's still a lot you can do with quite rudimentary keyboard skills just by recording in small sections and overdubbing the left- and right-hand parts separately.

Irrespective of how you capture your raw MIDI performance, though, I've noticed that a lot of MIDI controller keyboards (especially the cheaper synth-style unweighted variety) seem to favour higher MIDI velocity values, so one of the most common problems I find with the MIDI piano tracks I hear is that they cause the virtual instrument to be played too hard overall, resulting in a hard, brash tone that refuses to sit nicely in the mix without lashings of remedial upper-spectrum processing. As such, these days I often request the MIDI data for piano parts in songs I'm asked to mix so that I can turn down the MIDI velocity values *en masse*. I rarely do this by applying a straightforward velocity offset, however: I prefer to use a global velocity multiplier of between 0.9 and 0.7, thereby reducing the velocity value range as well as the overall values. This is because the velocity handling of some MIDI controller keyboards (and of the people playing them!) can be a bit haphazard, so making the velocity values a bit more consistent usually provides an ancillary benefit. In addition, I'll almost always turn down any obvious velocity spikes individually to avoid single notes that suddenly leap out of the balance undesirably — intentional musical accents excluded, of course!



The top screenshot here shows a common problem with many project-studio MIDI piano parts: the velocity values are simply too high. Multiplying all the velocity data by a factor of 0.85 in this case brought the velocities (and therefore the piano timbre) to a more musical-sounding place, as you can see in the second screenshot. However, it's also common for occasional isolated notes in a live-performed part to be unduly accented, and these are better reined in individually, as shown in the bottom screenshot.

Concerns about MIDI velocity values usually run a lot deeper than that in most real-world scenarios, though. (This stands to reason in a way, seeing as the two main variables of piano playing are when you hit each note and how hard you hit it — indeed, many people consider piano a member of the percussion instrument family for this reason.) If you've created your raw MIDI performance piecemeal, for instance, it's very easy to end up with ungainly velocity transitions between overdubbed sections, or to find that separately performed right- and left-hand parts don't balance appropriately against each other. This may demand your applying further velocity offsets to smaller MIDI selections, and I think it's worth attending to those edits before attempting any more detailed velocity contouring.

Velocity Edits

In terms of editing the velocities of individual notes, there are two general tactics I use to guide me. When it comes to evaluating the musicality and phrasing of piano parts, one of the best guides is simply to sing an approximation of whatever part you're scrutinising. The mechanics of physically playing a piano part can somehow desensitise the player to awkwardly realised phrasing, whereas singing is a much more instinctive form of expression, so comparing the two is excellent at highlighting the former's musical deficiences. And you don't have to be a trained singer, either — this techique will work just as well even if your voice curdles milk at 50 paces!

The second great workflow hack when refining your piano part's MIDI velocity data is to audition the part within the mix context and refocus your attention from musical concerns to simple issues of mix balance. Can you hear the musically interesting lines well enough? Are the upper registers dominating over the bass notes? Are any unimportant beat subdivisions being over-stressed at the expense of the rhythmic groove? Or are there certain notes that are causing unflattering frequency build-ups in combination with other instruments in the mix? This last question deserves special attention, because it's extremely common for some chord notes (often the root notes of the harmony) to be doubled sporadically by other parts in the arrangement, such that these pitches unduly dominate in the balance. Bear in mind, too, that a lot of less experienced keyboard players have difficulty balancing the power of their thumb against their fingers when playing oscillating or broken chord figurations, so this is another common source of localised MIDI velocity imbalances. From a mix perspective it's usually a lot better to adjust velocity values to spot-fix such moments at source, because trying to rebalance things with audio processing plug-ins downstream can be a bit of a nightmare.

The Quantise Trap

Most live-performed MIDI piano parts benefit from a certain amount of timing correction, in my view, simply because timing vagaries that pass unnoticed within the context of a one-time gig can become a bit distracting when heard repeatedly within a recorded production. However, I strongly urge you to avoid the shortcut of applying any kind of large-scale quantising routine. Let me explain why.

What you need to realise is that the constituent notes within any live-performed piano chord almost

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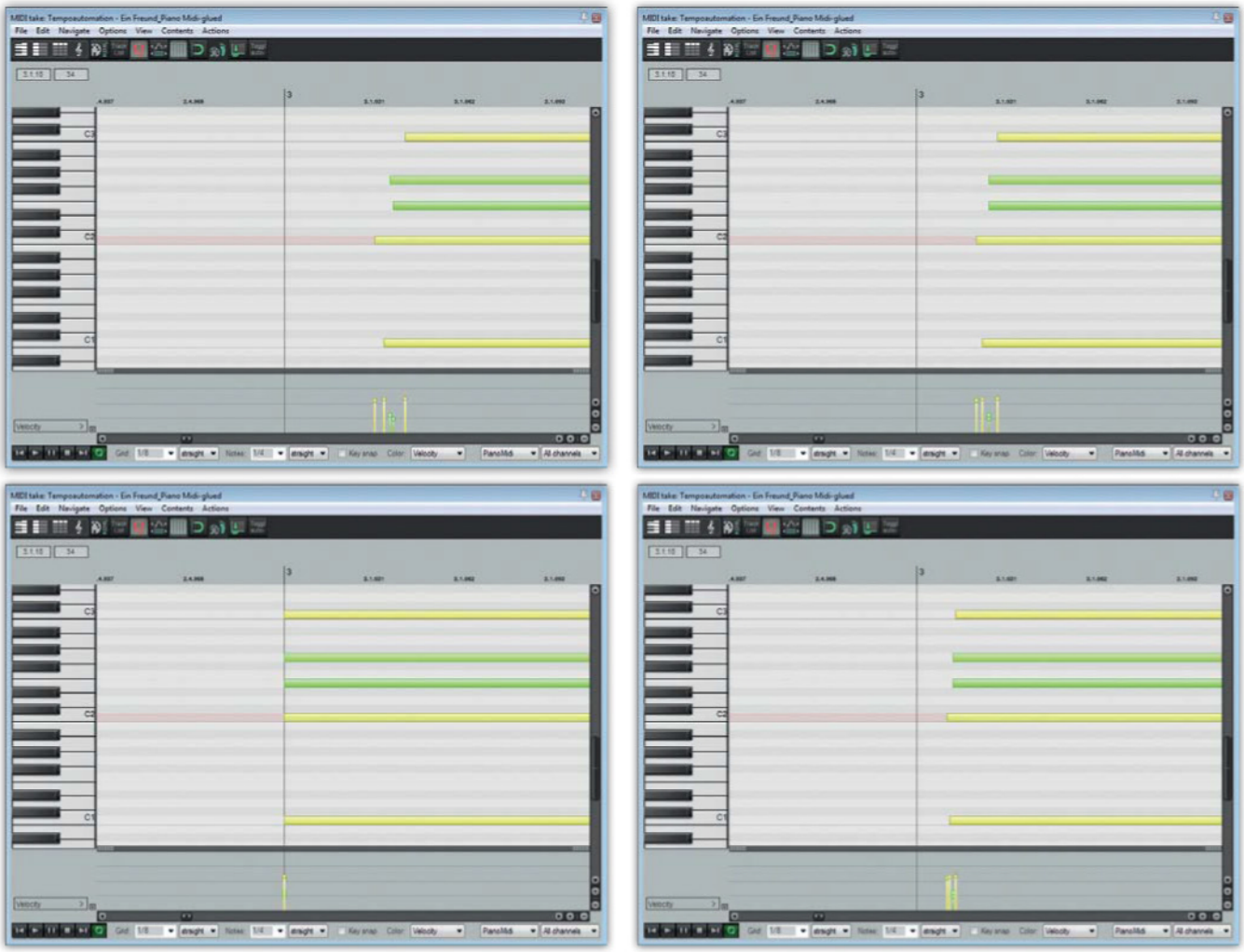


The top-left screenshot here shows the MIDI notes of a live-performed piano chord that's behind the beat. If you try to remedy this lateness with quantisation, the naturally splayed timing of the notes gets increasingly ironed out as the the chord is progressively pulled towards the metric grid — as you can see if you work your way clockwise through the remaining screenshots.

» never occur at exactly the same time. In fact, this time-splaying of chord notes is a crucial and integral part of what makes piano parts sound human, and it's frequently exaggerated for musical effect in many styles by deliberately staggering the note onsets from the lowest to the highest — creating an expressively 'spread' chord.

With this in mind, consider the situation where a quarter-note piano chord is played, say, a 32nd note late, such that all the naturally time-splayed chord notes fall behind the quarter-note grid line in your MIDI editor. If you try to correct the chord timing by quantising towards the grid line, you'll also 'squash' the natural time-splaying between the chord notes. In other words, the more firmly you quantise in this way, the more you dehumanise the performance. Modern quantisation functions offer plenty of fancy options like variable strength, adjustable quantisation 'window', groove templates, and so on, but I've yet to find any scheme that doesn't fall foul of the same essential dehumanisation problem when dealing with piano parts. And don't expect any kind of MIDI 'humanise' function to bail you out here either, because the more or less randomised timing variations those typically generate are a poor substitute for the human time-splaying that meaningfully responds to the musical context.

So what to do instead of quantising? Well, I hate to have to say it, but I reckon the best way to serve the music is to work manually, grabbing each wayward chord 'cluster' in your MIDI editor and sliding the whole thing forward/backward in time without interfering with the timing relationships between the constituent notes. Obviously you need to evaluate the efficacy of any timing adjustments by ear, but I find I can usually get into the right ballpark visually by deciding what the notional 'centre' of the chord is, rhythmically speaking, and dragging that to the metric grid. The location of a chord's rhythmic centre will clearly depend on how it fits into the desired groove in your music, but for me it usually seems to be about a third of the distance between the onsets of the first and last notes in the chord cluster. Clearly, if you encounter some chords that feel rhythmically too indeterminate in a more uptempo production, then slightly reducing the time-splaying can help tighten things back up.

Pedal Power

Another common shortcoming of project-studio MIDI piano parts is the sustain pedalling. This is something that non-pianists sometimes leave out completely, whereas most believable piano parts use the sustain pedal most of

Choosing & Using A Software Instrument

I've talked a lot about improving your MIDI piano parts through programming tweaks, but the MIDI instrument you're funnelling all that data into constitutes a vital part of the recipe too. It can be tough to choose the right MIDI piano instruments, though, seeing as there are so many contenders available on the market these days, ranging from the bundle presets in hardware keyboard workstations and software samplers, to highly sophisticated virtual instruments dedicated to fastidiously recreating specific piano models. So let me offer some advice here.

Above all, you need to find an instrument that suits the musical style and arrangement contexts you're typically dealing with. Pianos for classical-style scoring applications, for instance, will typically have a woodier and more conservative tone with plenty of rich low midrange, and while you might be able to press such an instrument into service (with some judicious 200-300 Hz EQ scoop) to serve a smooth jazz number or stripped-back singer-songwriter ballad, you'll struggle getting it to fit into any high-energy pop, rock, or EDM production, where hard-edged attack and a strong upper midrange tend to be more what the doctor ordered.

A second important priority is to find a patch with a sufficiently smooth velocity response. What I mean by this is that you should check that the timbre of the sound changes in a believable way as you gradually increase the velocity values of notes. Many of the piano patches in big sample-library bundles suffer from too few samples per note, for instance, with the result that you get obvious and distracting timbral changes as you cross the boundaries between velocity ranges. So, for example, you might find that velocities 50-99 use a single sample, so changes in velocity only adjust the volume of the note within that range; but then when you reach a velocity of 100, the piano's timbre suddenly changes radically, because the sample engine switches over to a much harder-sounding sample that's assigned to the 100-127 velocity range. This kind of thing can make it very difficult to play/program musical-sounding parts, so I recommend carefully vetting potential instrument choices on this basis. (If you're stuck with a slimline instrument with bad velocity switching, though, then one passable workaround is to restrict your Velocity values to a limited range to avoid any obvious timbral discontinuities, and then use Continuous Controller 07, Continuous Controller 11, or DAW fader automation to refine the part's level/phrasing beyond what velocity data editing can manage under the circumstances.)

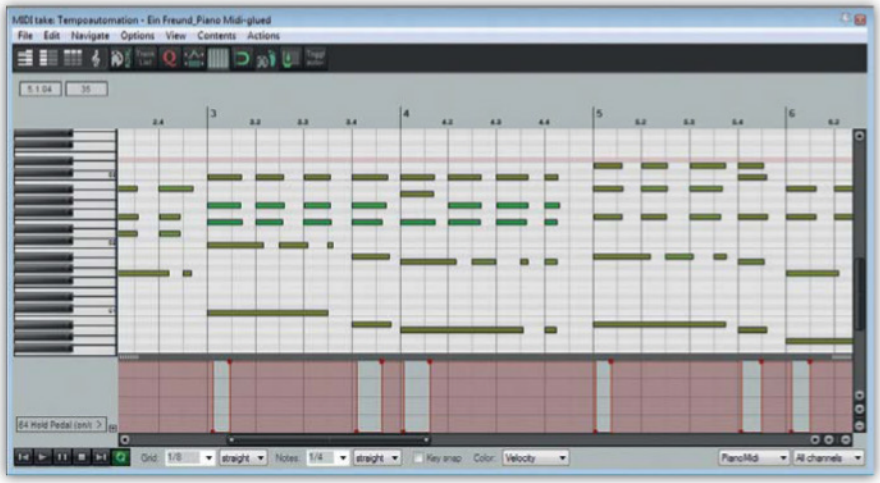
My third big wishlist item for any virtual piano instrument is that it simulates a real acoustic piano's sympathetic resonance in some way. You see, pressing a piano's sustain pedal in real life undamps not just the played notes, but all of the other notes too, which enriches the overall tone and also adds a subtle sense of coherence to the sound. What's more, even with the sustain pedal up, a significant amount of sympathetic resonance still occurs. A software piano that simulates such resonances tends to feel more like a single organic instrument than a bunch of sampled notes, and it'll also usually provide that additional lovely characteristic tonal 'bloom' that acoustic pianos deliver whenever the sustain pedal is pressed down.

Beyond that, there are plenty of instruments that go to inordinate lengths to emulate an instrument's mechanical noises as well, but to my mind those are mostly just unnecessary gimmicks. It used to be that we'd spend ages in studios trying to avoid capturing those sounds, so we're not used to hearing them that much within mainstream styles. And if you really do want the indie credentials of a creaky old upright, then frankly you might as well just record one, as that'll likely give you a much more unique and ear-catching result than any programmed instrument is capable of.

the time, even where the player would be perfectly capable of holding all the notes down without its assistance. The MIDI standard uses Continuous Controller 64 for this purpose, with values of 127 and zero corresponding, respectively, to the pedal being pressed down (ie. 'on') and released (ie. 'off'). With real acoustic pianos, there's actually a certain amount of range to the pedal's effect between these extremes, and some virtual instruments attempt to simulate this, but I wouldn't honestly bother with that — I reckon it's overkill for mainstream productions.

If you don't have much experience with using a sustain pedal, then the main thing to understand is the way the pedalling action typically relates to your chord changes. Broadly speaking, the normal scheme of things is that the pedal is pressed down to sustain each of the song's harmonies in turn, and briefly released to damp any conflicting notes of that harmony from ringing on just as the next harmony sounds, following which the sustain pedal is reapplied to sustain the new harmony. This gives sustain-pedal data a characteristic look to it during sustained harmonic playing, where the Continuous Controller value stays at 127 most of the time, but then hops down to zero for a moment just after each harmonic change. The critical decisions from a programming perspective, then, are:

- When the zero region starts: too early and the previous harmony won't transition smoothly into the new one; too late and the start of the new harmony will clash with the sustain of the previous one.
- How long the region lasts: too long and the sound may lack a sense of sustain; too short and the previous chord won't be sufficiently damped, again blurring the harmonic transition.



In the Continuous Controller 64 editing lane here you can see a normal sustain-pedalling data pattern, involving the pedal being released for a moment after each change of harmony.

Piano Pro

You can make a big difference to a MIDI piano part's musicality and believability by simply addressing velocity values, note timings and pedalling data using the tips laid out above. And if you'd like to hear audio examples of some of the things I've been writing about here, then visit https://sosm.ag/better-midi-pianos to have a listen for yourself.

Create drum machine-style grooves in a flash with Cubase's Beat Designer.

Beat Designer: drum machine-style pattern creation with a simple workflow.

JOHN WALDEN

Classic drum machines such as the LinnDrum or those in Roland's TR range helped to shape a whole smörgåsbord of electronic and dance music styles, but their influence didn't only live in the distinctive sounds they offered: the grooves of their drum patterns were also a hugely significant factor. The feel of these grooves came from the grid-based pattern creation process that lay at the heart of these machines — and with Cubase's Beat Designer MIDI plug-in you can emulate that workflow. What's more, with a few key tricks under your belt, patterns can be created really quickly. So, without further ado, let's explore Beat Designer's, um... beat designing options!

Basics: Sounds & Beats

Beat Designer is placed as a MIDI insert plug-in on whatever instrument track you wish to use. You'll need to load a suitable drum instrument, of course, and the bundled Groove Agent SE or a sliced drum loop within a Cubase Sampler Track both make good candidates. Beat Designer does include some good beat presets and, as I'll discuss below, a single instance can host up to 48 different patterns (four sub-banks of 12 patterns each). It's easy to switch between patterns too, and if you set a cycle region and let Cubase loop playback, you can audition your pattern creations as you work. Initially, though, Beat Designer opens ready for you to start creating from scratch, without any pattern pre-loaded.

The central part of Beat Designer's GUI is occupied by the pattern grid. Arranged vertically on the left side is the list of eight individual drum lanes and you can add/subtract lanes as required via the small '+' and '-' buttons on the far right of each lane. If the default drum selections don't suit, you can use the drop-down menu in each lane to pick the drum sound target from your drum instrument. You can also use the small speaker icon to audition each selected sound; it's simple and easy to configure.

You can add hits by either clicking on a single lane/step, or clicking and dragging to add multiple hits (for example, on the hi-hat lane). Where you click vertically within a step provides initial control over that step's velocity (high velocities at the top, low at the bottom). Hits are colour-coded based on velocity. Clicking and dragging up/down on a step lets you adjust velocity or, if you hold Shift while clicking/dragging, change the relative velocity up/down (until a step

You can configure Beat Designer's lane layout to suit your needs.

If you don't want to roll you own, Beat Designer includes a good selection of pattern presets to explore and adapt.

reaches min/max velocity values) for all active steps in a lane. You can also change the velocity on selected multiple steps by clicking/dragging vertically on one step, and then dragging sideways across other active steps before then moving up/down to set the velocity. Equally, you can add volume ramps up/down by repeating this same mouse movement while holding the Alt/Opt key. And, of course, clicking on an already active step deactivates it.

This whole process is really simple and obviously makes for very speedy rhythm creation, but there are further useful shortcuts on offer too. For example, holding Shift, clicking on a step within a lane and then dragging left/right allows you to move all the steps within a lane. You can also copy one lane to another by holding Alt/Opt and clicking within the lane label area, then dragging to the lane where you wish to place the copy. And, for some further interesting rhythmic experiments, hold Alt/Opt and drag left/right over a lane: this inverts all the steps (so active steps become inactive and *vice versa*).

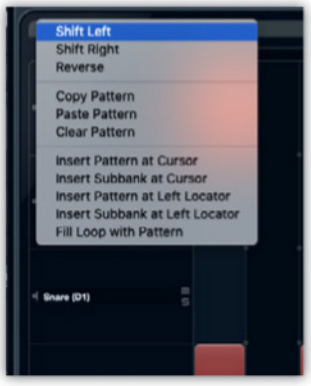
Step, Swing, Flam & Offset

The number of steps and the step resolution can be set on a per-pattern basis. Both settings are located top left of the GUI, and there's some fun to be had here, particularly if you create a pattern using a tried and trusted 16 steps/16th-note step-length format and then change the number of steps or pick a triplet step-length format. Equally, if you use two separate drum instruments, each with its own instance of Beat Designer but set to different step counts or step lengths, you can be rewarded with some very cool — and sometimes wonderfully odd! — rhythmic interaction.

Beat Designer can also apply swing on a per-lane level to every second beat in a pattern, or every third beat if using triplet steps (so bear in mind that the number of steps you use influences how swing impacts). For each pattern, you can configure two levels of swing, using the sliders with the diagonal icons that are found at the bottom right. For each lane, swing can be set to off or to one of the two swing amounts, using the appropriate diagonal icon at the right end of each lane. This can be a great way to add a nice groove to your hi-hats, for example, but more extreme swing settings on your hats, kick and snare also let you experiment with the more 'uncomfortable' rhythms used in some types of hip-hop or lo-fi music.

You can add one-, two- or three-note flams to any active hit — their timing and level are configured using the group of sliders located bottom left and set at the pattern level. If you hover the mouse over the bottom of any active step, a three-dot icon will appear and you can click this to add one, two or three flams to that step.

The horizontal sliders found to the right of each lane (next to the on/off buttons for swing) apply a timing offset for that lane, allowing you to push/pull the timing of the active hits relative to the main grid. This serves two main functions. First, you can use it in a corrective fashion; for example if the lane is triggering a sample that contains a short rise to its peak level you might want each hit earlier in time for that sound. Second, you can use it more creatively to create a rushed (ahead of the beat) or laid-back (behind the beat) feel to elements of your groove. Or if Dilla-style hip-hop beats are your thing, you can use it to explore just how 'off but not off' you can take your rhythms. As with the swing settings, offset settings are configured at the pattern level.



The Pattern Functions menu allows you to perform various operations in Beat Designer or export the MIDI patterns to your DAW tracks.

Pattern Performer

As mentioned above, a single instance of Beat Designer can hold up to 48 different patterns. These are managed using the four sub-bank and 12 pattern buttons, found at the bottom centre of the GUI. Slots that contain a pattern are indicated with a small circle icon, and clicking on a specific sub-bank/pattern button combination makes that pattern active for editing/playback. Dropping down the Pattern Functions menu (found at the top right) provides menu options for copying, pasting and clearing patterns. This makes it easy to create variations: just duplicate a pattern and then customise the copy.

This menu offers a couple of other useful functions. First, the top three entries let you shift an entire pattern left/right one step at a time or to reverse the whole pattern. Second, courtesy of the options at the base of the menu, there are a number of ways to copy the contents of a pattern, or all the patterns within one of the four sub-banks, onto a Cubase MIDI or instrument track, and once there you can further edit the MIDI using the Cubase's standard MIDI or drum editors. A particularly useful option is the Fill Loop With Pattern command: this inserts the current pattern as many times as required to fill the region between your left and right locators, and it's a super-quick way to get your initial Beat Designer drum groove up and running so that you can work on other elements of the track. (Once you've copied your pattern data to a MIDI track, remember to bypass Beat Designer, so as to avoid any double triggering of notes!)

If you prefer to create a performance by triggering pattern changes on the fly, enabling the Jump button (top right of the GUI) lets you use the four-octave key range from C1 to switch between any of the 48 pattern slots. Once Jump is enabled, the GUI adds a Now button that allows you to toggle between instant pattern switching or switching at the start of the next bar, and you can record the trigger notes to capture your performance (and, if you wish, edit them after recording).

Get Creative

You can create all sorts of beats using the various tools I've outlined above (inverting lanes, different step lengths, shifting lanes left/right, etc) but the real beauty of Beat Designer is its speedy workflow — if classic drum machine beats are what your project needs, then it is well worth taking the time to explore it. Meanwhile, just to give you an idea of the sort of pattern variations you can generate really quickly, I've included some audio examples on the *SOS* website: https://sosm.ag/cubase-0823.

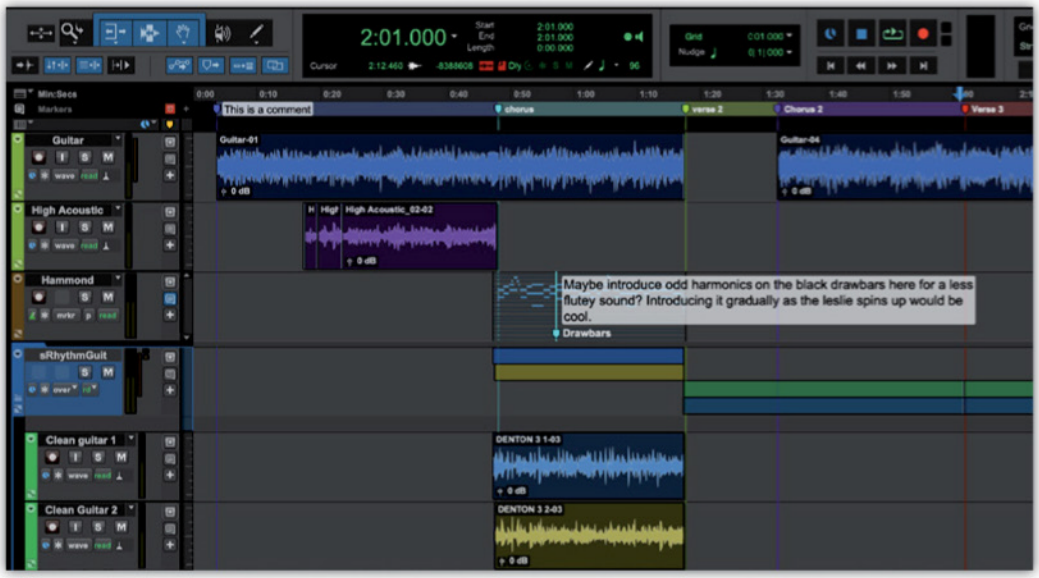
Navigate busy sessions with ease using Pro Tools' new Markers system.

JULIAN RODGERS

At the beginning of June, one of the longer-standing feature requests from the Pro Tools community was finally addressed when Markers received an extensive overhaul. These new features were previewed at the NAMM show in April and released in Pro Tools 2023.6, and now that I've had some hands-on time with them, I can say that they don't disappoint.

The headline here is that there are two new types of Marker: Track Markers and Folder Markers. The old timeline Markers, displayed on the ruler, are unchanged. Instead of adding additional Marker rulers, the new Marker types are displayed on tracks themselves, either within a new lane or on the main playlist.

Markers can now be added to any track type, but if they're added to a Folder track, a specific Folder Track Marker is created. These show up in the Memory Locations as a separate category of Marker. The difference between Track and Folder Markers is that Folder Markers display a line that extends down across the track lanes in the same way as a Marker on the Ruler does, except that in this case the line only extends through tracks that are members of that Folder.



The three new buttons at the right of the track header allow you to add a new Track/Folder Marker, select a particular track as Target for Markers created using keyboard shortcuts, and toggle between the Marker name and its associated comment.

Pushing Buttons

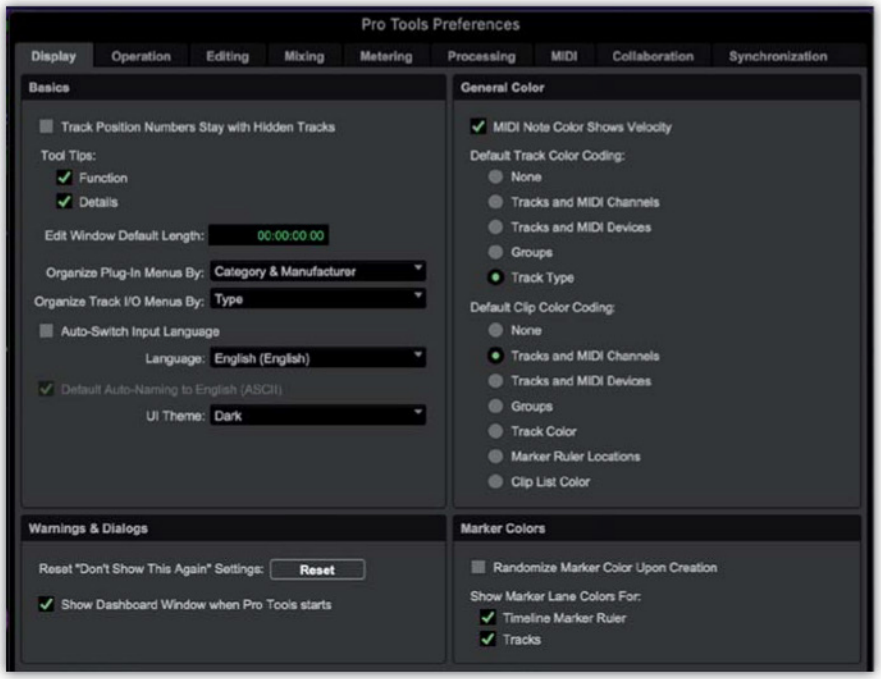
There are some new UI elements to accommodate the use of these new Marker types. The most prominent of these are the three new Marker buttons in the track headers. The bottom one is an Add Marker button, denoted by a '+'. Click this to create a Marker. At the top is a new Target button. This serves the same purpose as the Target buttons found in plug-in and send windows, allowing the user to direct an action towards a specific destination. In the case of Track Markers, this Target button specifies which track a Marker will be created on if the new Add Track/Folder Marker shortcut of Command+Enter (Control+Enter on PC) is used. The Enter shortcut for creating Markers on the timeline ruler is unchanged.

New Marker options in the Preferences window allow you to assign a random colour to each new Marker, and to display Marker colours for Track and timeline Markers.

Using this new shortcut, a Track or Folder Marker is created at the playback cursor position. If your timeline and edit selections are linked, as they usually are, then this will also be at the insertion point, but actually it follows the playback position, not the edit selection.

The third of these new buttons toggles between showing the Marker name and its associated comment. This greatly increases the usefulness of comments, by making it possible to display the comments field that has always been available in the Memory Locations Window on the timeline in the Edit window. These new Marker controls can be displayed or hidden using the View menu under Edit Window Views. They can be Option/Alt-clicked to hide them in the same way as other elements of the Edit window.

A significant change to all Markers is the use of colour. All Markers used to be yellow, and while there was the option to Always Display Marker Colours, this coloured the space on the ruler between the markers rather than the markers themselves. This preference is replaced by a choice of display marker colours in the ruler and/or on tracks, and a preference to randomise marker colour on creation. With this unticked, Markers will be created using the previously selected colour, which might be useful when you want to create markers that are related in some way. The colour of a new marker can be controlled using the 1-8 keys in combination with either Command or Option (Control or Alt on PC).



The Memory Locations window lets you sort Markers by colour.

The New Memory Location window now offers a choice of 16 colours for new Markers. As well as visually differentiating between Markers, they are useful because the Memory Locations window offers the option to sort by colour, opening up the possibility to use different colours to denote and group Markers.

Making Memories

Other changes have also been made to the Memory Locations Window. It's now fully resizable and has a new Track Name column, which displays the track on which the Marker resides and an accompanying icon to denote the track type.

These new Markers are simple to use, powerful and, importantly for a DAW as mature as Pro Tools, don't impose changes on those who don't want or need them. The operation of Ruler Markers is unaffected by their new brethren, but is added to in useful ways. The ability to colour Markers is welcome, as is the ability to display comments on the Markers ruler.

The Track and Folder Marker workflow is intuitive but really powerful. With Link Edit and Timeline and Link Track and Edit Selection enabled, you can use the P and semicolon keys in Command Focus to move up and down between tracks and then hit Command+Enter to create Markers exactly where you want them in a very intuitive way.

By default, Track Markers populate a new mini-lane underneath the main Playlist. There is a new Markers view available for the main track lane, but the super-useful shortcut Shift+U to toggle the Marker lane visibility is addictive. People using longer comments will find Markers view important, as the text of long comments wraps if the track height is large enough to allow it. The height of the dedicated Marker lane is small and fixed, and comments don't have the same text wrapping.

There are a couple of Marker-related settings in the Options menu. The first is essential if you want to control targeting of tracks for Marker creation. Target Follows First Selected Track is necessary if you want to click in a track's playlist or use P and semicolon to control track targeting with Link Track and Edit Selection as described earlier. If you prefer not to work this way, then an alternative shortcut is to use Shift+square brackets to move the targeted track up and down.

The second Marker-related Options menu item is Markers Follow Edit. This works in a similar way to Automation Follows Edit, and in fact the newly added Toolbar button is next to Automation Follows Edit underneath the Scrub tool. This option allows Track Markers' positions on tracks to be linked to the content on the track. There are times when you might wish for both types of behaviour, and this option allows that.

A useful modifier is to use Ctrl (Mac) or Start (Windows) while clicking on a Track Marker to snap it to the beginning of an Edit Selection or to the Insertion Point. Command+Ctrl-clicking on a Track Marker (Ctrl+Start on Windows) snaps a Track Marker to the end of an Edit Selection. Double-clicking on the space in between two Markers (Track, Folder or Ruler) creates a selection between them.

You've always been able to drag Ruler Markers horizontally along the timeline, and the introduction of Track and Folder Markers adds the ability to drag Track Markers vertically between tracks. Usefully, you can drag a Track Marker between tracks of any type, including from a track to a folder and even from a track to the Marker ruler. A nice touch here is that the Marker is constrained horizontally while dragging vertically, unless it is dragged a significant distance horizontally. At this point it becomes 'detached' and can be dragged any distance, including a distance smaller than the amount it was dragged before it became horizontally unconstrained. Horizontal dragging follows the behaviour of whichever Edit mode you are in.

Final Thoughts

Comments in Markers have always been available, but because they were only visible in the Marker Window, they were previously underused. When you created a comment in a Ruler Marker in a version of Pro Tools prior to 2023.6, it could only be viewed within the Edit Memory Location window. Having these comments available in context in the Edit Window means they can be used to make timestamped notes for collaborators, or for your future self as a reminder. In the past, people have used track comments for this purpose, but they aren't time-specific. Another oft-used but less than satisfactory workaround was to name clips or to create empty clip groups on an otherwise empty track to serve as unofficial additional markers. Track Markers with the option to Prioritise Marker Name vs Comments give the flexibility to label not only the time at which a comment pertains, but also the specific track or range of tracks to which it relates. If no comment is present on a Marker, its name will be displayed instead. Because Prioritise Marker Name vs Comments is a setting that can be enabled on a per-track basis, notes and snag lists can be worked through one track at a time. And a final workflow enhancement is the shortcut Shift+N to toggle comments on and off. In combination with Shift+U to toggle Track Marker lanes on and off, the rich information these new Markers can display doesn't have to get in the way when you don't need to see it. Nice!

DP's extensive Bounce options make exporting stems a breeze.

MATT LAPOINT

The audio production term 'stem' generally refers to a processed, self-contained stereo mix of an instrument or family of combined similar instruments. For example, a stem could consist of a solo trumpet or an entire brass section. Stem breakouts are usually dictated by a project's specific delivery requirements. In post-production, greater flexibility is often required because musical elements are mixed with dialogue and sound effects. In song production, stems are a common delivery format that allows for variations across distribution and deployment scenarios, such as licensing and live playback. Some mastering engineers will request stems from the mix engineer to create a final master. In addition to processing, stems usually include all rendered edits, pitch and timing adjustments, as well as volume automation and processing. Stems can be dry, with separate reverb-only stems being additionally created, or stems can be printed with all effects baked in. Through the use of bouncing tracks and outputs, Digital Performer 11 can bounce stems, tracks, busses and outputs, all from one window, all at the same time.

Templates

Bouncing is an essential and efficient way to finalise mixes and stems for delivery, especially for film and TV composers. Bouncing files for delivery has the potential to be a tedious and time-consuming process if projects and templates aren't set up correctly in advance. You should carefully decide what breakouts will be necessary for a given scoring project, and design and modify track routings before your writing process begins. When the time comes to send reference mixes, or final exports and stems, the routings and selections will be ready to go.

When designing a comprehensive DP template, it's important to know what kind of tracks can be included in a Bounce to Disk operation. Individual audio tracks can be bounced separately, and will include all track insert effects, soundbite edits, and automation. This is also true for aux tracks and combined instrument tracks.

Here's an important tip: audio tracks and virtual instrument tracks can be assigned to stereo busses that feed aux tracks. Because aux tracks can be bounced as tracks, this is a way to combine audio tracks and virtual instrument tracks into stem groups.

Screen 1: DP's Bounce to Disk window. In this example, aux busses have been set up in advance to produce stems with one click.

When using reverb plug-ins on different families of instruments, it's important to create separate reverb plug-in instances for each stem group. Otherwise, common-use reverb returns could bleed into multiple stems from unwanted instruments using the same reverb. Separate reverbs, created using aux tracks, can be bussed to their corresponding stem aux tracks so the audio and reverb are both sent to the same destination stem. Use the stem aux tracks for overall level control, EQ, compression and limiting as well, if necessary.

MIDI tracks assigned to multitimbral virtual instruments require separate bouncing for audio separation. Fortunately, most VIs have multiple outs that can be assigned to separate aux tracks.

Master fader effects are not included when bouncing tracks in the Bounce window. If master fader effects are an integral part of the signal processing, bouncing outputs will be necessary. Outputs can be both hardware outputs and DP internal busses. With busses, aux tracks are the best way to control the gain stage.

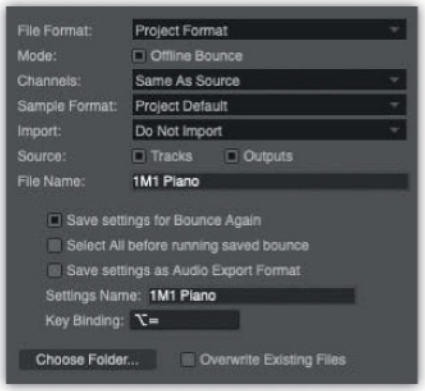
Making Selections

The bouncing features in Digital Performer are based on selecting MIDI and audio data in the timeline first. If data isn't visibly selected, it won't get bounced. Make sure track folders are open (expanded) to ensure all the track data gets selected. There are a few ways to make selections in DP. Click and drag a selection across single or multiple tracks, or use Select All inside various edit windows in DP. The Tracks window is quite useful when making bounce selections in DP (see Screen 2).

At the top centre of the Tracks window is the Selection Information strip. Selection start/end times are displayed by bars/beats/ticks. The Time Formats window in DP's Setup menu can change the time format, if desired. Using the menu to the right of the start and end times, selections can be based on Chunk Bounds, Remember Times, the counter position, and Saved Times. Markers are another useful way to create selection boundaries. First, Select All and then click the name of a marker in the timeline. Doing so highlights the data from that marker to the next marker in the timeline. Select all tracks you are ready to bounce. The Bounce window intelligently bases the track

Screen 2: Use the Tracks window and Selection Info strip to choose what to bounce. You can even save and load selections for instant recall.

Screen 3: The Bounce to Disk settings.



Screen 4: The Bounce Again settings.

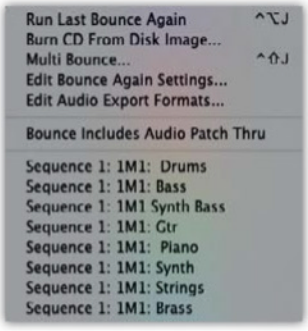
Screen 6: Multi-Bounce lets you create stems from multiple saved bounces, even across multiple DP projects, in one go.

sources on the selection made before invoking the Bounce window.

Formats & Other Settings

After selecting a combination of tracks and outputs for bouncing, choose File / Bounce to Disk and then use the left column to choose file settings (Screen 3).

Bounced files can be uncompressed formats such as Broadcast Wave and AIFF (either interleaved or de-interleaved), or compressed formats like MP3. The Project Format option is based on the file settings currently chosen in DP Preferences / Audio Files. Bouncing as a movie is also an option for both Mac and Windows platforms. Mode allows bouncing offline or in real time. The offline bounce speed varies based on the types of plug-ins and virtual instruments that are being bounced. The Channels setting provides options for surround channels (if surround bundles are being used in the project), stereo, and mono (with and without attenuation). Sample Format provides options from 16-bit up to 32-bit floating-point files. Import provides the option to import the bounced audio back into the project timeline. This option is only available for uncompressed formats. If this is set to Do Not Import, the default location of the files is the Bounces folder on the root level of every project folder. At the bottom of the column, Choose Folder allows for a custom location for the bounced files. Overwrite Existing Files is an additional option at the bottom to preserve or write over files with the same names. Source can be tracks, outputs or both. File Name is for naming the file and adds the output and track name as an extension to the chosen name. There are three additional tick boxes at the bottom of this column. The first two will be further explained below. The third option, Save settings as Audio Export Format, saves all of the upper menu choices (File Format, Channels, Import, and so on) as a preset that can be recalled from the File Format menu. Saved Audio Export Formats are global and available in all other DP projects.



Screen 5: Bounce Settings.

Bounce Again & Multi Bounce

Save Settings for Bounce Again is a powerful feature that allows you to bounce the project again and retain all necessary settings, including selections, tracks, outputs and file format. Use the Settings Name field below to name and create a Bounce Again preset. A custom keyboard shortcut can be assigned to invoke this preset again. Time-saving tip: after selecting a track and time duration to save as a Bounce Again setting, run the bounce, but then cancel the process right after it gets started. You don't have to wait for the bounce operation to be completed: all bounce settings are retained, even if you cancel. Select All Before Running Saved Bounce will perform a Select All command for the entire timeline boundary of a project (or active chunk) before bouncing.

DP's File menu contains a Bounce Settings sub-menu with options to Run Last Bounce Again, Edit Bounce Again Settings, and Multi Bounce (see Screen 5). The Bounce Includes Audio Patch Thru menu option, when checked, leaves track inputs active during a real-time bounce so that, for example, an external hardware synth could be included in a real-time bounce if routed to a stereo track set to Audio Patch Thru. Saved Bounce Again settings appear in the Multi Bounce window and in the Bounce Settings menu (at the bottom of the menu). The Multi Bounce window displays Bounce Again presets that were created in the currently loaded project. However, it also displays any presets that you have loaded (from other projects) while no project is open. So, if you have presets that you always want available, regardless of which DP project is currently open (or with no project open at all), then close all projects (keep DP running), open Multi Bounce and then click the Add button to load them from the desired DP project(s) on disk.

The Edit-related menu options allow for deleting and renaming saved settings. The Multi Bounce window lists saved Bounce Again settings created in the current project (see Screen 6). Remove deletes selected settings. Add provides a file dialogue window where other project files can be searched for and selected. Bounce Again settings are stored in project files and can be added to the Multi Bounce window. After selecting all or specific saved Bounce Again settings, hit OK and DP will perform the bounces.

Tip: the Multi Bounce can be accessed while DP is running but no project is currently loaded. Depending on the request, DP will open projects, make selections, and perform bounces, then close and move to the next bounce operation. A progress meter will appear giving feedback on the status of the Multi Bounce operation. Once completed, a report will pop up to list what was completed.

Bounce To This

Stem Bounce is a powerful and convenient method of exporting a combination of audio tracks, instrument tracks, aux tracks and outputs at the same time. Combined with Multi Bounce, Digital Performer provides the flexibility to quickly bounce stems, from the early stages of a project all the way to final delivery.

Routing for an extensively miked drum kit. Some drum tracks go directly to the main drum bus (21), while others are routed there via aux tracks for submixing. You can see that all of the individual tracks are in a group.

We continue our crash course in editing and mixing drums.

DAVID RICARD

Last month, I outlined some of the techniques and methods I use to organise and prepare live drum tracks. While those steps may seem a bit tedious, they are essential to ensuring an efficient workflow further down the line.

It should be noted that, even if working with live drums isn't in your immediate future, the techniques and concepts I'm sharing here are applicable to many other audio editing tasks.

Let's assume that you took my advice and your live drum tracks are now grouped, gain-staged, and in the correct polarity, and that you've created Track Alternatives for each of the different drum takes.

Auxes In Motion

While assigning busses and aux tracks to your drums may fall more under the umbrella of mixing, there are indeed some editing tasks that benefit from it. It's not unusual to have several tracks for the kick drum (in, out, sub), snare drum (top, bottom), toms, and ambience (room mics, mono overheads), so it makes sense to bus each of these groups to their own aux track. Processing the individual tracks is still possible, but now we have an easy way to process them as subgroups, and to quickly mute and solo them.

Start by bussing the drum tracks to one master drum bus. Select all of your drum tracks in the Main window, right-click (or Control-click) on any of the tracks and select Create Track Stack. Select Summing and hit Create. If all of your tracks were already bussed to an aux track, Logic preserves this in the Track Stack output.

Newer versions of Logic Pro allow you to create nested Track Stacks that would do most of this work for you. Simply apply the same steps to each group of tracks as we did for the main Track Stack.

While we are at this stage, let's create those aux tracks I mentioned. In the mixer, hit Control+N multiple times to create between eight and 12 auxes. We won't only be using these aux tracks for our drum routings: we will also use some for effects, reverb and parallel processing. As you add each one, they will pile up on the right side of the mixer. Let's Shift-click to select all of the new aux tracks. While holding down the Option key, click on the input of the leftmost aux and select a bus (preferably one with unused busses after it), and Logic will assign each aux's input consecutively.

We now need to get our aux tracks into the Main window and into our drum stack. With all of the auxes selected in the mixer, hit Command+T. Now go to the Main window and drag the aux tracks into the drum stack. Once they're in the stack, Logic should have changed their outputs to that of the stack.

Select the three kick drum tracks, for instance, and change their output to the input of the aux you'd like to use for your kick subgroup. Carry on in this fashion with any combinations that work for you. How you choose to route the individual drum tracks is a personal choice. Of course, not every track needs to go to an aux; they can go directly to the main drum bus.

As for processing, add any effects you wish to the remaining auxes. Reverbs, ambiences, compression, distortion... all fair game!

Start Making Sends

I like to have sends to these busses in place on each drum track, whether it's a candidate for such effects or not. Then I simply disable the send knob if I don't need it. Start on the leftmost track and add sends for each of the effect auxes you've created. You can dial in some rough levels and disable the ones that you don't want to use.

Now go to the Settings tab and choose Copy Channel Strip Setting. We certainly don't want to paste all the settings into the other tracks — after all, we've just set up our routing and added gain plug-ins for flipping polarity — but we do want to copy the sends. Go your next drum track, click on the Settings tab and choose Paste Sends Only. Repeat for the rest of your drum tracks.

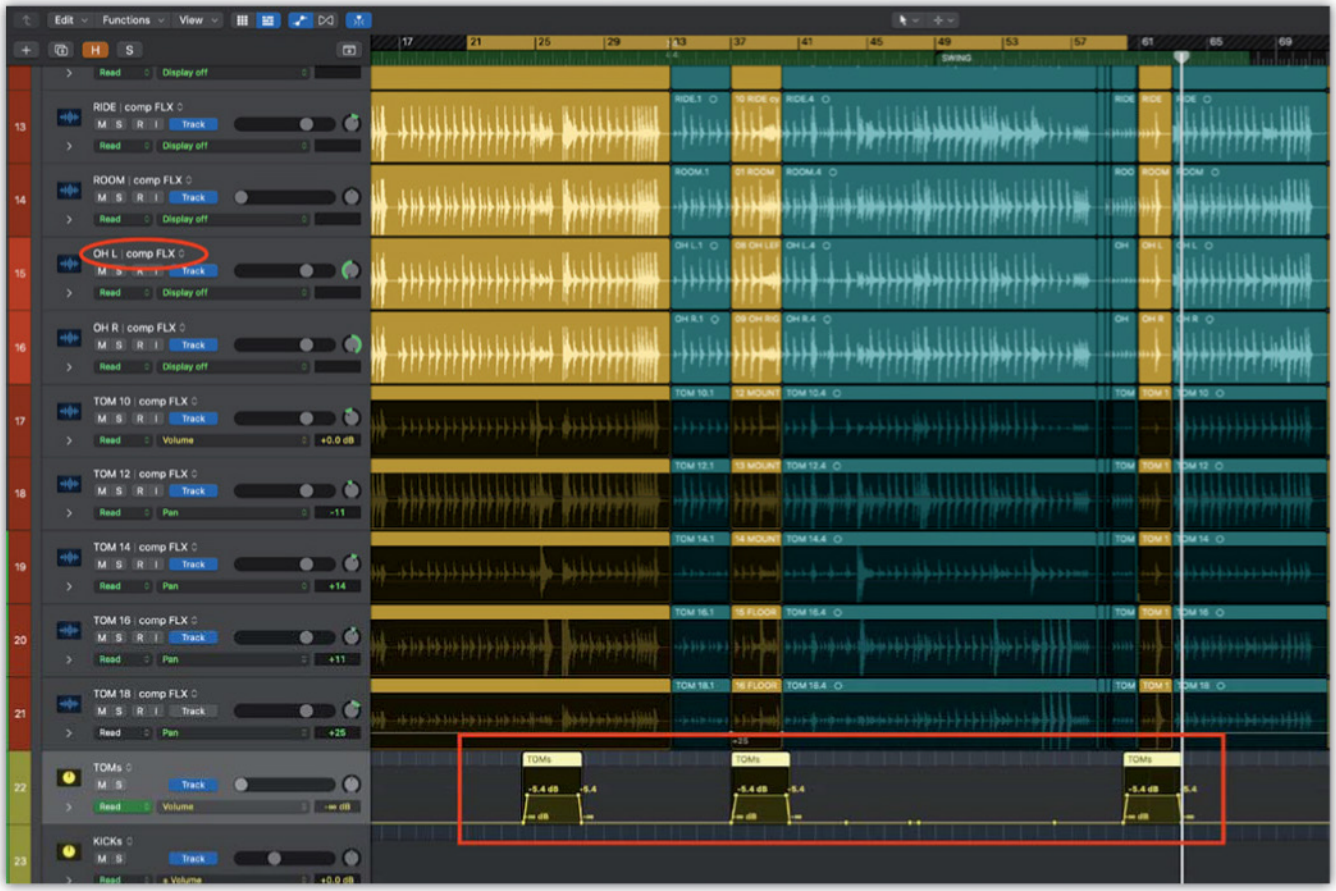
Tom Tom Club

In the next instalment of this series I will be discussing an array of techniques for 'cleaning up' your tom-tom tracks in Logic. However, I wouldn't recommend putting them into practice until after you have a composite edit (including Flex edits) that you're satisfied with. There is one exception, though, and that's my region-based automation tom hack.

Typically, the tom tracks sound great on the actual tom hits. The problem is, the spill when they're not playing tends not to sound great. Compounding this issue is the fact that any EQ and compression that you might want on the tom hits can make the spill sound even worse.

My quick-and-dirty tom mixing hack cleans things up pretty well for this stage and, in some cases, may be all you need for the final mix. Start by moving your toms bus close to your toms tracks in the Main window. Get a ballpark level

Once you've created aux sends for one drum track, you can use the Settings tab to Paste Sends Only onto the others.



A completed drum comp. Because the different takes are colour-coded, you can easily tell which take each section was taken from. At the bottom, you can see my method for easy tom level automation.

that works well when the toms are actually active. Make a mental note of where that level is on the bus's fader. Now pull the fader all the way down.

Locate a section of the song where the toms are active and, with the Pencil tool (Shift+T), draw a region on the toms aux track. Position the region so it starts a beat or two before the first tom hit and extend its length so that it ends a bit after the last hit on the sequence.

Now hit A to enter Automation mode. The track should default to showing volume automation. Draw nodes at the beginning of your region, a beat into it, a beat from the end, and at the end itself. Then grab the line between the four nodes and bring it up to the level you made a note of earlier.

Now you can select this region and paste it anywhere in the timeline where the toms are active. The automation gets copied with the region! You can get pretty detailed in dialling in the perfect amount of automation for each region, but the point of this technique is to simply achieve a working balance while you continue tweaking your masterpiece. Even better, it should work with any of the other drum takes!

What I love about this technique is that I can visually identify where there's activity on the toms at a glance. I sometimes even hide my tom source tracks after this stage to tidy things up a bit.

Talking Edits

In Part 1, I discussed loading (or recording) each drum take onto a Track Alternative. Now we need to put together one performance from the various takes.

Logic names each Track Alternative automatically, but I would rename them so they make more sense to you (not just now, but when you revisit the session later!).

Here's how I approach assembling a comp. First, colour the regions of each take differently. Then identify the take that is closest to being perfect. Create a copy of this take by choosing Duplicate from the Track Alternative drop-down menu. If this isn't visible, make sure Track Alternatives is ticked in Track Header Components. Since the tracks are grouped, this only needs to be done to any one of the drum tracks to affect them all.

Rename the duplicate something like 'Comp 1'. Now, however much damage we inflict on these drum tracks, you always simply switch back to the original Track Alternative.

As you audition the tracks, just concern yourself with the overall performance; we will be getting into Flex quantisation next time. For now, we just want the best takes for each section in our comp.

When you arrive at a section that could be improved, set the Cycle range around it. Use the key commands for Next/Previous Track Alternative (or switch it manually) and see if the section is better performed in any of the other takes.

If it is, copy the section by selecting any of the drum regions, then going to Edit / Cut/Insert Time / Cut Selection Between Locators. Copy the regions (Command+C), go back to your comp Track Alternative, and move the playhead to the beginning of the edit point. Make sure you have the topmost track selected, and paste (Command+V).

If you've never comped a drum performance together, you may find it surprising how little finessing it takes to mix and match the takes. Usually all that's needed is a little adjustment of the beginnings and ends of the regions (crossfades are rarely necessary) to make a seamless comp.

We explore how Studio One's Snap, Grid and Quantize settings interact.

ROBIN VINCENT

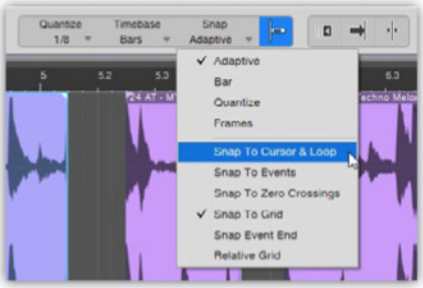
A couple of workshops ago, I asked for suggestions for future articles. The first request to land in my inbox asked if I could cover snapping, as the sender found it absurdly complicated. They're not wrong; there's plenty of room for confusion in terms of terminology and what settings are actually in charge of the Grid. So, armed with a magnetic cursor, let's dive straight in...

Grid Pro Quo

Snap is all about getting events in a tempo-based project to land in meaningful positions when you move them around. It persuades tools, the song position cursor and the edges of clips to align themselves to a subdivision of the Grid you choose, which can be bars and beats, seconds or frames. If you are working on a free-flowing, non-tempo-conformist piece of music, then Snap probably won't be of great use to you. But for those of us who are slaves to the rhythm, it can be extremely helpful.

When we talk about Snap and Grid we are essentially talking about the same thing, but the naming is a bit inconsistent in Studio One and it's easy to get confused. When you are setting the Snap value you are actually setting the Grid, and what you are doing in your project is Snapping to that Grid. However, the visual lines on the arrange window, which you could easily assume are representative of the Grid, are there regardless of your Snap/Grid settings and are actually governed by the Timebase settings. The fun thing is that Snap, Grid and Timebase, as well as Quantize, can all define the Grid — but they don't have to. I'm sure we'll get the hang of it.

Snap isn't a directive; it's more of a strong suggestion. It acts like magnetism to pull your event or tool onto the Grid. I find that I can almost feel it, haptically in my mouse, when that pulled snapping occurs, but that's just me. The strength of the pull depends on what the Grid is set to relative to the level of zoom you are working at. So, if Snap is set to Bars and you are zoomed in to a couple of bars, then you can move an event around freely between the bar lines, and you'll only feel the pull when you are very close to the start of the next bar. If you zoom out, or set the Grid to a much higher subdivision (ie. a shorter period), then you'll feel the pull of the Grid everywhere, and your event or tool will automatically leap from Grid line to Grid line.



By default, Snap behaviour is set to Adaptive, which means that events will be magnetically attracted to Grid subdivisions according to the Timebase settings.

One top tip is that holding down the Shift key temporarily suspends Snap, allowing you to freely move what you need to wherever you wish to move it.

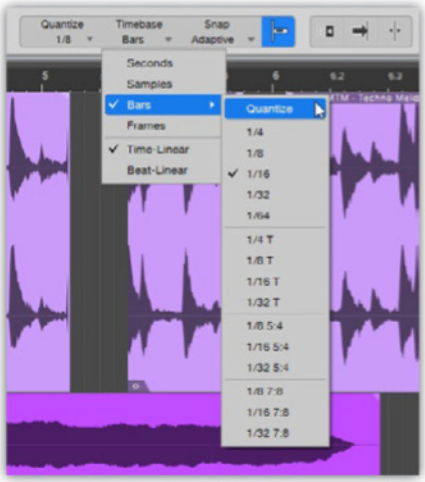
Let's get into the settings. The Snap/Grid settings are located with the Quantize and Timebase settings in a shared toolbar at the top of the main window.

Adapt & Overcome

The enigmatic Adaptive mode is the first and default setting for Snap. It shifts responsibility for the Grid onto the Timebase, which makes sense because we can see the vertical lines of the Grid right there on the timeline. So why isn't it called Timebase rather than Adaptive? Well, because Timebase is most commonly set to Bars, which punts the Grid responsibility further down the road to Quantize. So, in Adaptive Snap, the Grid is derived from the Timebase's use of the Quantize settings. It makes you wonder whether an overhaul of the Snap-to-Grid system might be long overdue...

However, it is more nuanced than you'd think. The Timebase can have its own Quantize settings that are different from the main Quantize settings, and so knowing which Quantize engine the Timebase is using can be vital. The important thing to know about Adaptive mode is that it's very good at following what you're actually doing with your music, because as you zoom in and out of your project, the Grid will adapt to something appropriate. Adaptive mode works best if you try not to think too hard about the Grid and just assume everything is going to be fine.

If you're cynical about the uncanny way Adaptive mode can follow your way of working, then you can be more assertive and select the Snap / Bar option. However, this only works if the Timebase is also set to Bars; otherwise, the snapping doesn't happen. It also has to be the main Timebase. You can't sneak in a secondary Ruler, set that to bars, and expect it to Snap; snapping always follows the main Timebase.



The Timebase options. When you select Bars and then a subdivision, Adaptive Snap will adhere to whatever subdivision you've chosen.

Further complicating things is the fact that you can't actually set the Timebase to be based on a single bar. Well, you can, but to do that, you have to set the Timebase to Quantize and then set Quantize to 1/1. Regardless of whether you set the Timebase to its own Quantize setting or to the main Quantize setting, the snap will then only pull into a full bar, not the Grid shown by the vertical lines.

So, when you set Snap to Bar, it really means it.

Quantize

If you want a bit more granularity then you can shift Grid responsibility over to the Quantize settings. The Quantize setting governs the Grid you can use for recording and placing notes, and goes all the way down to a hemidemisemiquaver, or 64th note. You don't always see the vertical lines for every subdivision; it depends on how zoomed-in you are. But regardless of the zoom level, the Grid is always followed by the Snap setting.

If you were working with video, you'd set Snap to Frames, so that your events would be in sync with the video frame divisions. There's no leeway given between frames; the Snap is always directly to a frame. Of course, the Timebase must also be set to Frames for Snap to work. The frame rate is set in the Song Settings and is usually dictated by the video footage.

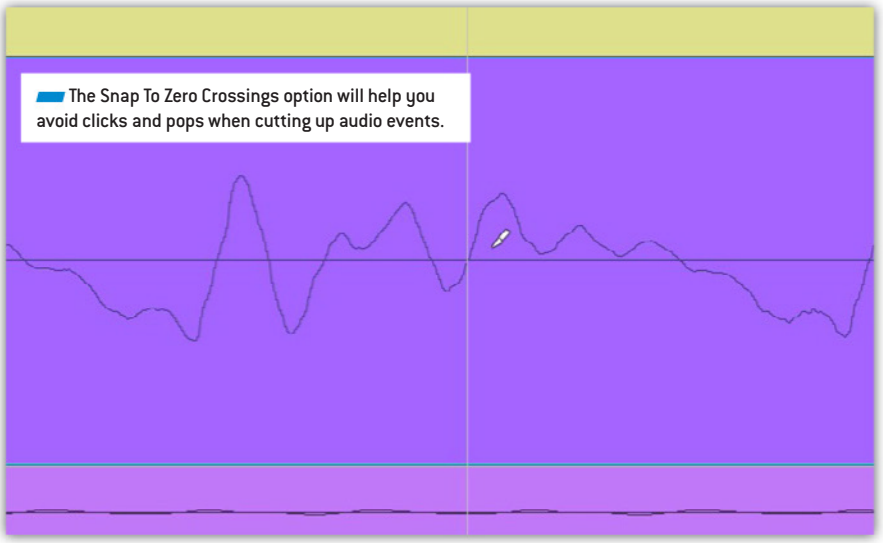
Snappy Days

Something quite neat about these Snap options is that they continue to work even if the rest of the Grid is off, and so become useful in more freeform projects. I should also point out that there are two different Snap enablers. Toggle Snap has its own button on the toolbar, and this turns the entire Snap/Grid engine off so that no snapping of any kind occurs. The other option is found under the Snap menu as Snap to Grid, which can be ticked or not. Snap to Grid only enables or disables the snapping of events to the Grid, not snapping as a whole, so the Snap options listed below will continue to work even when Snap to Grid is off. No possible room for confusion here!

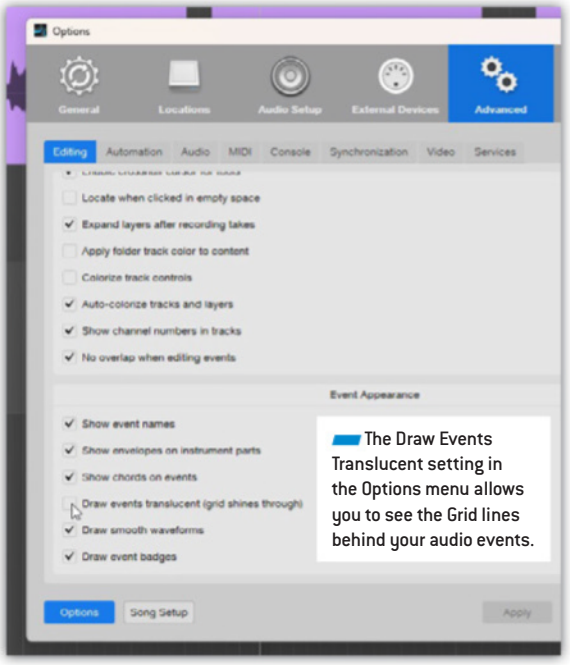
- **Snap to Cursor and Loop:** This makes the song position cursor and the left and right loop markers super-magnetic and snappable.
- **Snap to Events:** Events themselves become very snappy.
- **Snap to Zero Crossings:** If you are editing an audio event, you'll often want the cuts to occur at points where the waveform crosses the centre line — 'zero crossings' — to avoid unintended pops or artefacts at edit points. This is particularly useful when you are heavily zoomed into an audio track and you want to make precise cuts; your edits will always be right on zero crossings.
- **Snap Event End:** You may have noticed that when moving an event, it's the leading edge that does all the snapping. With this option enabled it will also snap on the back end of an event.
- **Relative Grid:** This is the option you need when moving off-Grid events around, for instance if you're editing an expressive performance with timing subtleties that you want to retain. The event's position will always remain the same relative distance from the Grid without actually snapping to it, allowing you to move it elsewhere in the song without losing its feel. If you can't get your events to snap to the correct place, then this option is probably on by accident.

Now that we know our way around the Grid a little bit, here are a few tips that make good use of it:

- **Split at Grid:** If you'd like to slice up some percussion loops or audio tracks into individual hits, you can make use of the handy Split at Grid command that you'll find hidden under the Event menu. In this instance, the Grid is set by the Quantize value, not the Timebase or Snap setting.
- **Grid Offset:** Have you ever wondered why we have to put up with odd-number bar counting? If you want a four-bar loop, where do you put the left and right markers? You put them on 1 and 5, or 5 and 9, and so on, which is a bit counterintuitive. Well, if you want to even up your loop selections, Studio One has the ability to offset the timeline. Go to Song / Song Setup, and you'll find an option for Bar Offset. Set this to -1, and your timeline now begins at zero, and all your even-barred loops sit on even bar numbers.
- **Unquantised audio snapping:** One of the new features in the 6.1 update was the introduction of Event Sync Points. These are to deal with the problem of musically significant moments inside events not always being right at the start or end of the event, meaning that Snap to Events or Snap Event End doesn't quite cut it. Right-click the audio event and tick the Sync Point box. A yellow diamond will appear, and you can drag it within the event and place it exactly at the point where you want it to Snap. Now you can move or copy your unquantised audio to different places in your project, and it will stay connected and in sync with your Grid.
- **Translucent events:** It's not always easy to identify timing issues in your recorded audio events. You might well be snapping the front and back ends to the Grid, the cursor or other events, but what about the performance in between? We could use the Grid to show us where the audio timing is getting a bit creative, but you can't see the Grid lines because the events are a bright, solid colour. Well, there's a box under Options / Advanced / Editing called Draw Events Translucent; this allows the Grid lines to faintly shine through the events. Now we can see if our waveform display is sitting correctly on the Grid.



The Snap To Zero Crossings option will help you avoid clicks and pops when cutting up audio events.



The Draw Events Translucent setting in the Options menu allows you to see the Grid lines behind your audio events.

INSIDE TRACK

SECRETS OF THE MIX ENGINEERS

Callum Marinho

The success of Noel Gallagher's *Council Skies* owes much to his resident engineer Callum Marinho, who not only tracked but mixed the album.

PAUL TINGEN

The first album to be recorded at Noel Gallagher's Lone Star Studio in North London, *Council Skies* reached number two in the UK charts. A return to Gallagher's more traditional singer-songwriter style after the electronic-music influenced *Who Built The Moon* (2017), it attracted widespread critical acclaim. It also respresented a huge career boost for engineer Callum Marinho.

Originally a drummer, Marinho's interest in recording led him to enrol in a Tonmeister course at the University of Surrey in the mid-2010s. "It's mainly classical, but there is a pop side, and because I am more into that, I ended up doing a placement year at Monnow Valley [*a residential studio in Wales*] as an assistant engineer. Bands come in for up to eight weeks to do an album, and I'd live and work with them. That was a big learning curve, obviously. I had a great time, graduated, and went back to Monnow Valley to work as their in-house engineer immediately afterwards."

Lone Star State

Tom Jones and his producer Ethan Johns visited Monnow while Marinho was there, and he ended up with an assistant engineer credit on Jones' 2021 album *Surrounded By Time*, which went to

Photo: Matt Crockett

Callum Marinho is resident engineer at Lone Star Studios in London.

'Council Skies'
Written by Noel Gallagher
Produced by Noel Gallagher & Paul Stacey

number one in the UK. "It was during that time that I also did a few sessions with producer Paul Stacey, who has worked a lot with Noel and Oasis. I eventually left Monnow and moved to London to start freelancing and one job I did was as an assistant with Coldplay in their studio, doing drum tech-like things. Three years ago, when Noel's studio was nearly complete, he was looking for an in-house engineer, and Paul recommended me to Noel. I've worked full-time for Noel ever since.

"The studio was pretty much ready to go when I arrived, but there were some teething problems, particularly with the monitoring. We had a couple of ATC SCM25As because Noel did not want big monitors, but they did not sound particularly good. So Noel, Paul and I got some ATC SCM150 ASL Pros and placed them on stands to see if they would work. Again there were issues, so we worked with Chris Walls of Level Acoustic Design, who designed the studio, to find a solution. We ended up soffit-mounting the 150s. It sounded great, so we now have 150s in the wall and Yamaha NS10s as nearfields, without a sub."

Noel Gallagher returns to a more familiar singer-songwriter approach with *Council Skies*.

Other equipment at Lone Star includes a vintage EMI TG12345 desk, similar to the one used by the Beatles in the last stage of their career, which Gallagher acquired in Australia a long time ago. "We use that desk to record all the guitars, going through the desk's mic pres, almost always with heavy compression on those channels. We also have 24 Neve 1064 mic pres, which are used for drums and vocals, and anything else that doesn't need the character that's added by the TG.

"There also are some Neve 2254 [*compressors*] here, which we tend to use for tracking bass; a couple of LA-2As that are always used when tracking acoustic guitars, and a UREI 1178, which was used on Noel's vocal across the album. The signal chain on his vocals normally starts with a Neumann U67, or a Shure SM7 if he sings really loud. We often have both mics on him at the same time, so we can later choose which mic to use and when. The mics go into the 1178, and then an AnaMod ATS1 tape simulator.

"The AnaMod is a Paul Stacey thing. He's got quite a few of them, and he takes them everywhere. They're the last thing on the chain, just before it hits the converters. We're trying to add the essence of tape, because we don't have tape machines. On drums it makes a clear audible difference on transients. It's quite subtle for non-transient things, but when it's on everything, you'll hear a difference when you listen to the whole thing. After that we go into Pro Tools. We have Avid HD I/O and the HDX system, and work with a 96kHz sample rate."

Underground Music

Work on *Council Skies* started early in 2021. "Coming out of the pandemic," says Marinho, "it was just Noel and myself during the demo stage. We recorded many demos. The formula was basically Noel playing acoustic guitar and singing to a tempo map. We'd then listen and judge what worked, so we could map the songs out as a rough guide. He'd then put a demo acoustic down, add a guide vocal over the top, followed by maybe a tambourine or a shaker.

"We did a lot of arranging. We sampled drums in the live room, and I made different sample kits out of those, one of which has a really good drum kit as a starting point. Noel would go over to a MIDI keyboard and play the kick and snare how he felt them, and then he put hats down, and cymbals and rides. After that he would stick some bass on it, and I'd play MIDI strings or MIDI piano.

"When we came to doing the actual album, we were thinking that we were going to start recording from scratch with Noel's band. But it's funny, we got so carried away with these demos that we ended up keeping a lot of demo stuff.

"Noel and I spent a lot of time flicking through presets on soft synths, for inspiration. There is a preset on the Arturia »

The live room at Lone Star, where much of *Council Skies* was tracked.

» simulation of the Synclavier, and Noel decided that it sounds a lot like the London Underground, when the trains slow down and echo around the tunnel. That's why he called that sound 'the tube'. It's on several tracks. Depending on the octave you play it in, you get different harmonics that ping out and fly around.

"I used outboard during demo'ing and tracking, and I'd then print these treatments. One unit we used a lot was the Publison Infernal Machine 90, which has a load of really good reverb presets. It doesn't sound like anything else. We used that on almost every track. Noel got it when we started to demo things, and was like 'Let's play with that and see what's in it!'

"Many of the reverb settings are really good. There's one called Cask which is a short, small room thing, and it also has different rooms, a plate, various halls, and at the end is something called Sound Hoarder, which is basically an infinite decay reverb that brickwall limits, so it flatlines on this wall of reverberating sound. The Publison is really fun. It works great on vocals, also with that Sound Hoarder preset.

"We also have an Eventide H3000, and an EMT 140 plate, which is in a big cupboard. Noel was really into that, so that started the sound on the record. There are several ethereal delays and reverbs on vocals. In any case, whenever we're tracking, I always have a bit of 140 plate on Noel's vocal, because it's a reverb that sounds good and it's not a plug-in, so there are no issues with latency. It's just Noel's vocals going through the 140, and coming back via his headphones. At the end, I'll record the 140 in on a separate track."

The members of Gallagher's band High Flying Birds overdubbed their contributions with Gem Archer contributing guitars, Chris Sharrock drums and Mikey Rowe keyboards. Paul Stacey added more guitars, bass and keyboards, and there were string and brass sessions at Abbey Road Studio 2. There also are tons of guest musicians, with eight backing vocalists, three percussionists, and Johnny Marr on guitar. The Pet Shop Boys, and the Cure's Robert Smith and drummer Jason Cooper, worked on remixes.

Serving The Song

An unusual aspect of *Council Skies* is that it bucks the current trend for music to have

Photo: Mike Banks / www.recordproduction.com

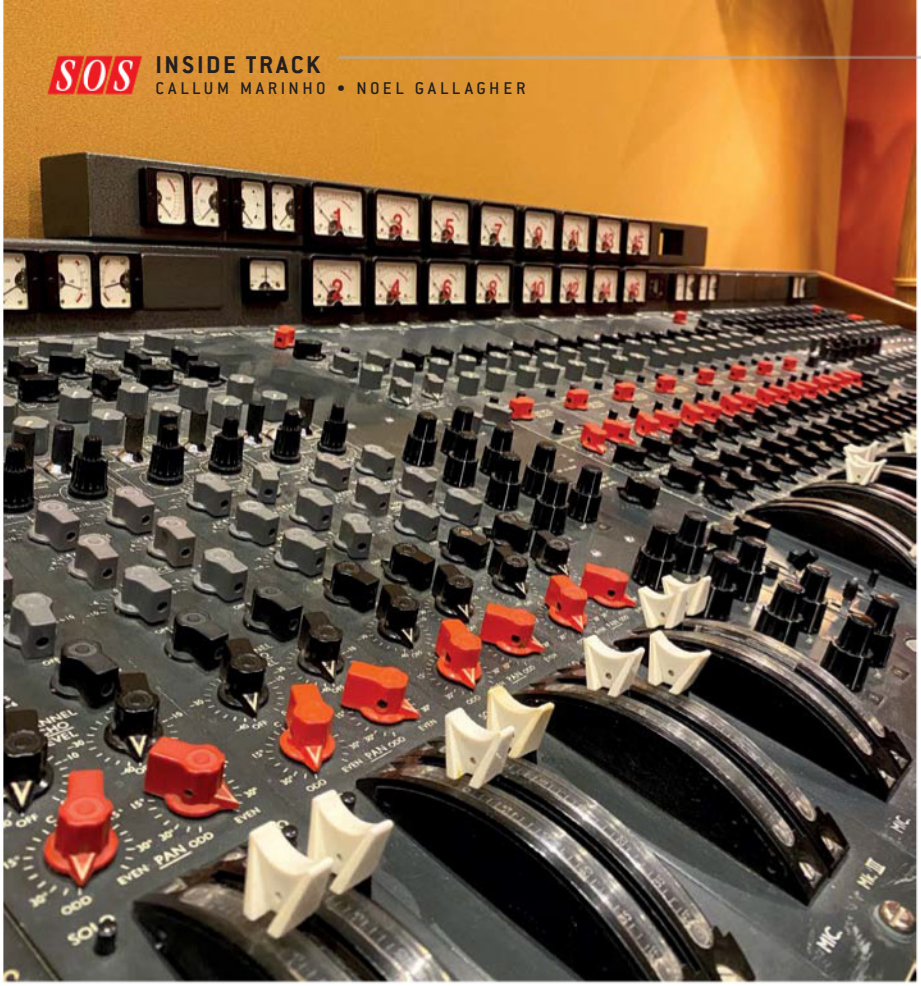
tons of low bass. “Noel always had an idea of how he wanted it to sound. He doesn’t like the bass being particularly loud. The main thing for Noel is always the song, and that should be what everything contributes towards. I guess in some other things that I’ve worked on a large amount of bass does accentuate the song. But I don’t think that would have suited this record.

“Also, we have a 16-channel SSL Matrix, which we use as a glorified monitoring station. Noel likes to be able to solo and change the levels of the different elements in a song himself, and have them where he wants to hear them. This happened from demoing all the way up to mixing. Noel doesn’t want to get involved in Pro Tools, so we needed to find a way to split things out from Pro Tools that allows him to balance things the way he wants them, without having to ask me every time to do it in Tools. So we ended up with him at the SSL as part of our workflow.”

Because Gallagher and Marinho had already developed advanced rough mixes with a clear sonic identity, the final mixes were a matter of fine-tuning rather than reinventing. “The reference was the rough mixes of the demos that we had done. We carried on in that vein. That’s why my final mix sessions don’t have anything crazy and there aren’t 10 plug-ins on every channel. Once it went from the rough mix to the actual mix there was no point rewriting the rulebook, and coming out with something completely different. It was very much just: take that, and iron out the little issues that I hadn’t spent so much time on.

“This album was very much a case of getting the sound right on the way in. There was no mixing through outboard, other than a couple of times when we reamped things using the TG, when things needed more juice. We did that with Johnny Marr’s guitar, squashing it through the TG and using EQ to tweak the presence and make sure it sat well with everything else. We also reamped some soft synths through the Neve mic pres to get them a bit more exciting.

“Other than that, it was just working with what we had recorded. I have all the aux tracks that I send to the Matrix at the top of the session, in green. On the title track they are drums, Abbey Road percussion, bass in »



The centrepiece of Lone Star Studio is a vintage EMI TG12345 console.

» mono, acoustic guitar in mono, percussion, Hawaii guitar, electric guitars, Johnny Marr, fuzz guitar in mono, strings, keyboards, Abbey Road horns, several aux tracks of backing vocals and one with lead vocal. The stereo output of the Matrix goes through a Neve 33609 compressor and an AnaMod, and back to Pro Tools, and I'd print that.

"My main mix moves in Pro Tools were to make a few adjustments and add some plug-ins to the tracks in Pro Tools, and I'd throw some plug-ins on the Pro Tools master bus. I was just making small tweaks, listening a bit more clinically on headphones and other spaces, trying to work out if anything stuck out, anything needed tweaking, being a bit more surgical with EQs, particularly in the midrange, and riding the levels of different bits coming in and out of the song."

Going Large

Callum Marinho's mix session for title track 'Council Skies' is gargantuan: the base count of 134 tracks includes three folder tracks for the Abbey Road strings, horns and percussion sessions. When they are unfolded the session would balloon to many more tracks. There are 16 drum tracks, six bass tracks, 25 percussion tracks and two sets of vocal tracks: a block of 19 tracks of main vocals, and a further set of backing vocals. The majority of tracks have no plug-ins at all, and most of those that are present belong to the Abbey Road strings and Abbey Road horns, one Pick Bass bus track, and the main vocal section.

Marinho says of the Abbey Road strings: "I've got the Avid 7-band stock EQ on it, with a high pass at 150Hz, and a couple of 3-4 dB cuts around the 400-500 Hz range. Things that are recorded in Abbey Road Studio 2 always have quite a resonance around that 500Hz range, and can take up too much space in that area. Then there's a UAD Pultec MEQ-5, which is just dipping a bit more at 500Hz, because it was still annoying me. I used the Oeksound Soothe 2 because when the violins played there was a bit too much of 3kHz, and Johnny Marr's guitar is prominent in that area as well. And then I had a UAD 33609 [*compressor*] plug-in to just level the strings out, because they were quite dynamic. Funnily enough I'm also using a Waves Abbey Road Chambers plug-in!

"All bass tracks are sent to that Pick Bass aux track, and I'm treating it with a UAD LA-2A to control them a bit, and a UAD Pultec MEQ5 pulling out a bit of 200Hz, plus a UAD Harrison 32 for a high-pass around 40Hz. We have two Universal Audio UAD-2 Satellite TB3 Octo units, so we always have enough processing power. I also added a Soundtoys Decapitator for some harmonics. This is probably an example of why there isn't masses of sub-bass in these songs. The sub comes more from the kick.

"I have the Avid EQ and the SPL Transient Designer on the kick. It was recorded with an AKG D12 or D112 on the inside, and a [*Neumann*] FET U47 on the outside, so I had two kick tracks. The Transient Designer pulls out some sustain, while the EQ pulls out 5dB at 100Hz. There were hardly any other treatments on the drums. Halfway through tracking I put a UAD 33609 on the drums bus, but in general, the drums for this track are incredibly open. The whole record is very open. I was a little bit concerned, because obviously with modern music there are the

The rare Publison Infernal Machine 90 supplied many of the vocal effects.

Soundtoys' Decapitator plug-in provided some grit on the bass guitar.

loudness wars, and when I went from these rough mixes, which were very open, to the final mixes, I had to find a bit more level, without limiting things too much, and changing the dynamic range.

"The main lead vocal has the UAD LA-3A and the SSL E Channel, taking out quite a bit of honk. I duplicated it, to create a parallel right next to it, and added the Soundtoys Devil-Loc for grit. It's at 1 out of 10, but for the Devil-Loc that's already smashing it! This lead vocal is actually quite different from a lot of the other vocals, because we recorded this with an AKG C28. We had to find a way of getting it more present. It was a bit more of a struggle to fit them in than the vocals recorded with the [*Neumann*] U67.

"These two vocal audio tracks have sends to a number of aux tracks, like one with the UAD SPL Vitalizer because we wanted to bring the vocal forwards. Others go to slap and delay auxes with the Soundtoys Little PrimalTap. There are also prints of the outboard H3000, Publison and EMT 140 plate. I ran the BVs through the EMT plate as well, but that's on a separate print track. I'm running the backing vocals through the same Little PrimalTap slap and delays as on Noel's lead, so they're in the same space.

"At the bottom of the session are the Strings, Horns and Percussion folder tracks, and then my mix bus. This is where it comes back in from the Matrix, having gone through the Neve 33609 and AnaMod. Sometimes the AnaMod is too much on the mix bus and I don't use it, but the 33609 is definitely my favourite mix bus compressor. I added another one on the Pro Tools mix bus! First there's a Neve 1073, then the 33609, the BAX Mix and the Pultec EQP-1A. It's all UAD. After that there are a Sonnox Oxford Inflator and a Massey L2007 mastering limiter, to add some volume. The L2007 is doing 1dB of limiting at the end, in case there is anything that is finding its way through the gaps. It mainly affects the drums, because there's no limiting or heavy compression on them.

"The last three tracks in the session are reference tracks. One has the latest rough that I sent, and the other two are mixes of other album tracks, so I can hear what it sounds like in comparison. I want to make sure they're all in the same ballpark. Many of the tracks on the album are quite different, and I tried keep them somewhat in the same space, so the record sounds cohesive."

TalkBack

Kabeaushé

WILLIAM STOKES

Kabeaushé is an alter ego of 29-year old artist and producer Kabochi Gitau. A formidable and wildly dressed character, Kabeaushé produces an eccentric blend of pop, hip-hop and electronica on his latest album *The Coming Of Gaze*. Although Kabochi is originally from Kenya, his career owes much to Nyege Nyege: a Kampala, Uganda-based collective, festival, community studio and record label. Nyege Nyege founder Derek Debru has brought dozens of underground East African producers, artists and DJs to audiences across Africa, Europe and beyond, and it was he who invited Gitau to travel to Kampala for a residency at the studio.

"There wasn't really a space in Nairobi, or a festival, where you could play and where people would completely understand you," explains Gitau. "My perspective, coming from East Africa, is that, like, you're either on this side or you're on that side. You're either making gospel music or you're making secular music — and if you're making secular music, you're probably talking about women. There's not a lot of music that makes you go, 'Oh! There's a bit of texture, a bit of aggressiveness!'

"The textures and the roughness and all the grit that's in there, all the clipping, that's something that I'm learning to enjoy more now than I than I have before. You can look at other, bigger studios and think, 'Oh man, it'd be so cool if I got that, or if I got that...'. But then you sit down with the reality and you realise: you've got this laptop, you've got this AKG mic, you've got this Behringer soundcard and you've got Fruity Loops. So what you're gonna do with it? Are you gonna make this music, or not?"

At the moment I can't stop listening to
Lots of stuff from the 1970s and '80s, particularly what was happening in France and Italy. I love, love, love Ennio Morricone and composers like Piero Umiliani. Ennio Morricone has albums upon albums upon albums, and everything sounds so different. One minute he's making something for, like, a spaghetti western, and the next minute he's making a love album. It's insane to me. And it's so minimal at the same time. That stuff makes me freak out.

I also just discovered this lady called Kate Bush. I did not know she existed. I did not. She has a song called 'Babooshka'. Ahhhh! It's the best song of all time! I was basically scouring the Internet trying to find sick music videos. When I found her I was just like, 'Who is this lady?' She has this playful, theatrical way in which she makes music, and I adore her. So I just went and downloaded all her albums. That's what I'm on now.

The project I'm most proud of
Oh, man. My next one! I can't say much more about it at this point. But I'm really, really excited about it.

The first thing I look for in a studio
What I look for is just a space — it doesn't even have to be a studio, it could be anywhere — where I can be secluded, and I can make as much noise as possible. I want somewhere where I can be as weird as I like without having to be super self-conscious of who's passing outside or what's happening around me. That's going to take me out of my element.

Necessity is the mother of invention. When I started I didn't have lots of equipment, and whatever you have is what you make do with. Sometimes you get tracks that will have a lot distortion on it — and then you try to take out the distortion, you try to clean it up, and it just sounds weird. I use an AKG dynamic mic from the '90s, and it picks up a lot of plosives and lots of noise and stuff! But it's the only mic I've found that sounds perfect for my voice, to the extent that I don't have to do so much processing on it.

The person I would consider my mentor
Derek [*Debru*] from Nyege Nyege. He's just as insane as me. I sent him an email saying, 'Hey, man, I'd really love to play Nyege Nyege festival because it's really, really cool.' I was new at it and had just quit my job. So I was, like, really still trying to figure out: what is this music thing? What's really my thing? So I sent him the music and he just sent back a really short email saying I should come for a residency. That was it. I had sent him a long-ass email, like, a really long email! And then he just responded, 'Hey, homie, this is cool. Are you down to come for residency?'!

So he flew me down to Kampala and we hung out. We played the album through a bunch of times and started playing around with ideas about the best way to release this music. I was coming in with just whatever I knew — I don't know as much as he does, of course. I was coming from a place of listening to people like Pharrell, without so much knowledge about music from the '80s, except tiny bits and pieces that I got growing up. So I was bringing, you know, what's happening in pop culture now, what's urban. And he'd say, 'OK, so here's this other side that you never knew of.' And we started, like, mixing that together.

And that's when we started to piece together a character, just taking references from all different things, and then making it African. I was like, 'OK, cool. Now we've found it.' Even the process of making the costume was about a whole month. It was a case of constantly saying, 'Let's go thrift shopping!' It's two different people: the person I am onstage and me offstage. It's sort of an antagonistic character.

Derek is just as fantastical as me, if not more. I don't know if I've met anyone as interesting as that man. He really has just been through everything. He's lived

everywhere. He does the craziest of things. You'll be hanging out with him and you'll suddenly find yourself in a king's palace, somewhere in the deep ends of Kampala. And you're thinking, 'How did I end up here?' It's very weird. Things just happen when you're with him.

My go-to reference track or album
When it comes to like drums and bass, I'm gonna say a lot of the stuff that Mike Dean does. He's really, really good with drums. Kanye's drums really knock hard. And Travis Scott, his drums really knock hard. The way the synths sound I also really love, like, that man's processing is insane. The way he does his vocals so everything's so 'at the top', but the drums knock super hard and the bass is super heavy. I don't know anyone who processes drums the way that he does. He does it in this way where to some extent it's as if it clips — or at least, it sounds clipped — but it just falls so beautifully.

My top tip for a successful session
I'm just gonna say: make sure you leave it all on the mic. Make sure you leave it all on the mic. That's it.

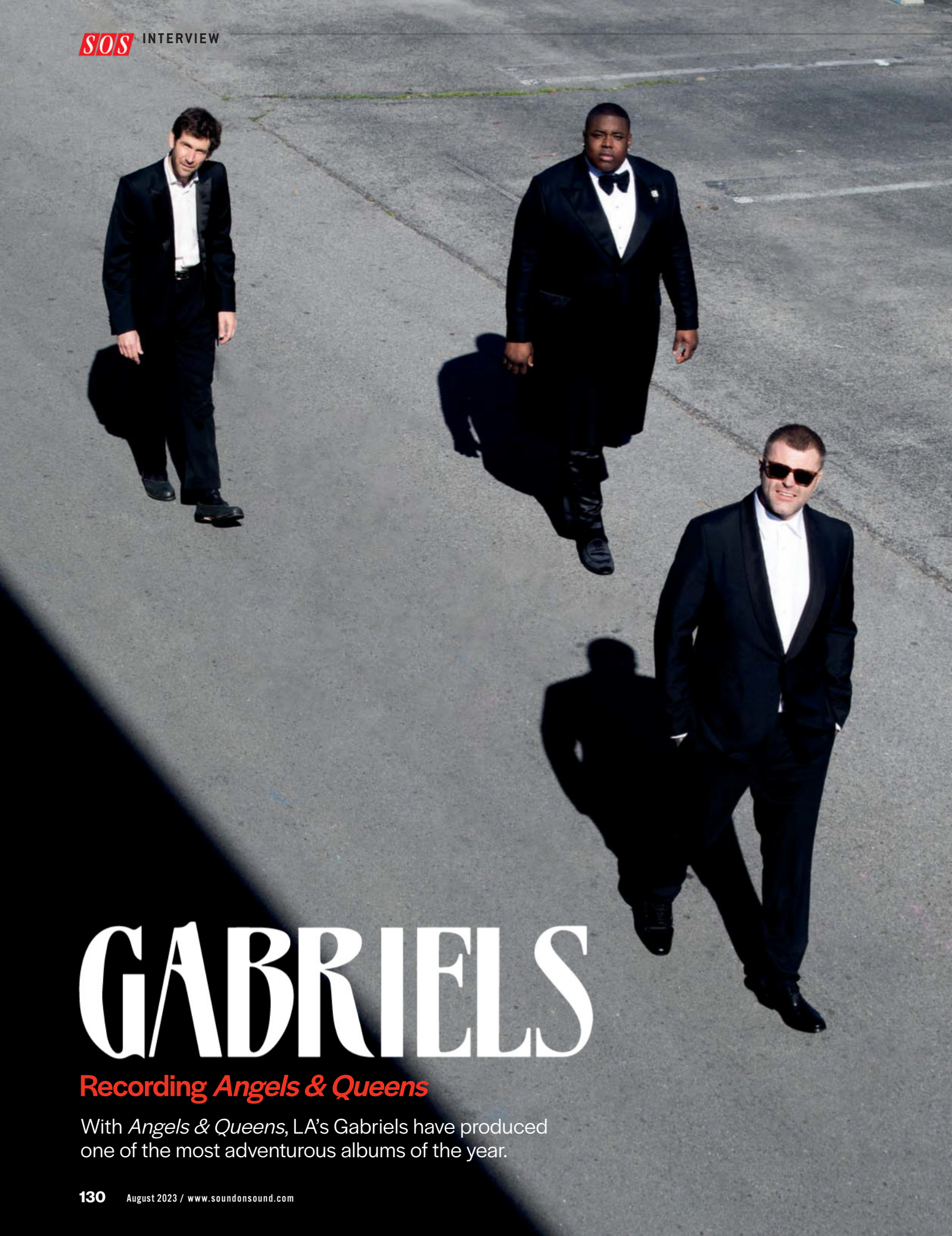
The studio session I wish I'd witnessed
I think just any Michael Jackson studio session would be great to be there for. He had this very sweet way of talking, a sweet way of expressing things and bringing out a song, but at the same time could be aggressive. He's, like, right smack in the middle of the spot between aggression and sweetness. And I love the fact that it was, like, melodies first. And then they would build on top of that. And the string sections... That entire *Off The Wall* album, man! It sounds very different to what he was doing with the Jackson 5, and it feels like it's the first time he's really starting to experiment. And Quincy is coming from his very knowledgeable background, actually writing music. You just have this kid who's like, 'I really love this music, and I can sing.' Fuse that all together and you get something that we still listen to now. And it still sounds very fresh. Twenty years from now, it's still gonna sound like it was made about two days ago.

The producer I'd most like to work with
Well, I feel like Quincy is too old now! No. It probably would be someone I'm really into now: this dude called Gaspard Augé from the band Justice. He just released

an album called *Escapades*. And the first song on that record is called 'Force Majeure'. And when you listen to the sounds, the synths on that song, the drums, it sounds to me like what Queen were doing, but he's sort of taking that and bringing it into the modern day. He's mixed, like, Daft Punk with what Queen were doing with their synths, all those '80s power synths, and then sort of infusing that with a really funky groove. I love how he thinks. I love how he's executed that song. And I think he sounds interesting enough to want to play with. He sounds like he's in tune with what's happening now, versus what was happening in the '80s. And it's also very cool that he's French. So he'll have listened to lots of that stuff I mentioned from the '70s and '80s! So yes: probably Gaspard Augé from Justice.

The part of music creation I enjoy the most
I love the whole thing. I can tell you the part I don't enjoy, which is mastering, because then you have to do the whole back and forth with the mastering engineer, like, 'Don't take all the energy out of the record! Let the 808s just bounce! Let them breathe! Don't don't mess around with the vocals! Just leave them as raw as they're sounding, because that's what the record is!' And then they will try and clean up all that stuff and you gotta go through all of that. That's my least enjoyable part. But I do enjoy the whole thing. From the minute I make the vocal on my phone to when I start recording, I love the whole thing.

The advice I'd give myself of 10 years ago
'Good luck.' That's it. It could work out, it could not work out. At least you did it!



GABRIELS

Recording *Angels & Queens*

With *Angels & Queens*, LA's Gabriels have produced one of the most adventurous albums of the year.

TOM DOYLE

Beginning with the 2020 release of their debut EP, *Love And Hate In A Different Time*, LA-based retro-futuristic soul band Gabriels have attracted widespread attention and famous fans, including Paul Weller, Harry Styles and Elton John — who called the EP "one of the most seminal records I've heard in the past 10 years". Now the trio, comprising singer Jacob Lusk and multi-instrumentalists Ari Balouzian and Ryan Hope, have finally released their full-length debut album, *Angels & Queens*, following a seven-track *Part One* taster issued last year.

The three all come from very different backgrounds: Compton-raised Lusk was formerly a gospel choir leader, working with the likes of Diana Ross and St Vincent, while Balouzian was a film and TV composer, and Hope, a Briton originally from Sunderland, was a music video and commercials director who'd moved to California when he was signed to Roman Coppola's agency The Directors Bureau.

Baked In

Gabriels first pooled their talents in 2016, when Hope and Balouzian were working together on music for a Prada commercial and invited Lusk to get involved. Since then, the group's music has maintained the same atmospheric, filmic quality, referencing classic soul and gospel recordings while at the same time employing modern, hip-hop-styled production methods.

"That's always been something we've liked," says Balouzian. "There's such a lack of things that are real nowadays. I feel like that's been lost a lot in modern music, where you don't feel people playing together.

"But, instead of doing it in a traditionalist sense where it's like, 'OK, this is a gospel blues album,' or something, it's like: How do you do that, but make it kind of fresh and exciting? I feel like we have those elements of hip-hop production, where they're sampling stuff that's all baked in together. With Gabriels, we could really have fun with the production and treat the songs like they were almost samples."

"It opens it up," says Hope, "because you're not reaching for a song in the conventional way. With our writing process too, it's very much like you're trying to paint a picture."

Desert Rock

The preliminary sessions for Gabriels' first recordings took place at Ryan Hope's home studio in the city of Palm Desert, East of Los Angeles. "I was making techno, really," he explains. "So I had a Roland 909, 808, 707, 727, 303. I had a Vermona modular drum machine. I had a Roland Jupiter-6, I had a [*Sequential*] OB-6, a Roland SH-101, a MicroKorg, a Yamaha DX7, an Oberheim Matrix 1000...

"I just had a shit ton of synths and drum machines, and I was buying and selling them. Whenever I got a cheque, I bought something, and whenever I needed to pay the rent, I sold something! Ari used to come out all the time. And basically, we used to just nerd out with all of the gear."

"He had this setup and then I would start moving stuff out there," says Balouzian. "I had a Coles [*4038*] mic and, eventually, we got a Neve 1073 channel strip. We just used those on everything. Vocals, strings: we would kind of get these little scraps and ideas. Ryan would have drum beats, modular synth stuff. I'd play keys over them, put strings on, and then we'd build stuff from there."

Sounwave: "Quantise is putting you in a box. If you learn how to actually program with no quantisations, that will put you in a whole different feeling."

Logical Moves

Up until this point, Hope had been using Ableton as his preferred DAW, but Balouzian encouraged him to switch to Logic. "I wanted to learn Logic anyway," he says. "It was a lot easier, because we were sharing files, for both of us to use the same thing. It's better for arranging songs too."

One of the pair's early breakthroughs was the creation of the brilliantly lopsided drums, claps and piano loop that forms the basis of one of *Angels & Queens*'s standouts, 'The Blind'.

"I'd been doing this for years, where I'd just put a mic into a [*Boss*] loop pedal and build these kind of weird drums," says Balouzian. "It was hitting the timpani, clapping, just one snare hit, all this stuff. And then I hit these chords, and that's all baked into the loop."

These handmade sessions continued at Balouzian's home studio in Burbank, where he would often layer string lines, playing all the parts himself, most notably on the original EP version of the strident soul/pop track 'Love And Hate In A Different Time'.

"I was doing it as an experiment," Balouzian says, "testing out my RCA 77 [*ribbon mic*], which I'd just got. I was like, 'What does that sound like if I use it on strings?' That more kind of thin string sound just ended up becoming a big part of the song."

Wave Hello

When Gabriels subsequently signed to Atlas Artists/Parlophone Records, and after they had written and toured most of the material for *Angels & Queens*, they decided to get a co-producer involved for the album, approaching Kendrick Lamar collaborator Sounwave (see box).

"The reason we were happy to bring him in is because we fucking love what he does," Hope enthuses. "The Kendrick albums are absolutely incredible. And it just feels like when you listen to them, you can feel the searching in them. That's very similar to the ethos of what we're doing."

"I had no idea that they were fans of mine, but I was a huge fan of theirs," Sounwave laughs. "The few EPs they dropped prior, I was just playing all the time. As soon as we got in the studio, it was an instant connection. Like, our energies matched. I'm always trying to push the envelope. So I don't like to stay in one box."

Back Tracking

The Gabriels album sessions began at Sunset Sound in Hollywood, before moving two miles Southeast to Conway »

Gabriels: Ari Balouzian (left), Jacob Lusk (rear right) and Ryan Hope (front right).

Co-producer Sounwave (front) and engineer Ryan Nasci behind the Neve 88R console at Conway Studios.

» Recording Studios. Live rhythm parts played by various drummers, including Trevor Estes (Midnight Sister) and Rico Nichols (Kendrick Lamar, Kanye West), were augmented by rhythms programmed by Sounwave using his Akai MPC Studio. "It's the closest I can get, in a portable, to the MPC2000 and 2000XL," he says. "It has the same functioning, same feeling, same vibe. The MPC, that's home for me."

The chief engineer on *Angels & Queens* was Ryan Nasci (Dua Lipa, Harry Styles), who was happy to employ minimal miking for the live drums at Conway, often simply positioning a single RCA 44-BX ribbon mic in front of the kit. "It's not like a rock setup, for sure," he stresses. "It's definitely more leaning into the jazz world. Then Sounwave just has his sample libraries that he's accumulated over the years. I'll loop the song, and he'll run through samples and grooves. And when he has something, he's like, 'All right, lay it in.'"

"Sounwave did an incredible job of helping us and encouraging us," says Balouzian. "There was that kind of freedom of him working with his MPC on our tracks and bringing his input. Y'know, drum beats we weren't really expecting. He'd be like, 'What do you feel like you need on this song?' Or he'd have done some work on one of the demos, and we'd say, 'Oh, let's maybe recut the bass and the rhythm stuff to your drums to fit your groove a little more.'

"At the end of 'Remember Me', he did this drum beat that he wasn't sure about at first. He was like, 'Is it too Wu-Tang or something?' And we were like, 'There's no such thing!' So we recut the drums and bass to those kinds of things."

Inspired by the late J Dilla, Sounwave has disabled the quantise function within his MPC. "I'm always pushing the swing," he says. "Quantise is putting you in a box. If you learn how to actually program with no quantisations, that will put you in a whole different feeling. Like that vibe alone is just so natural, so weird, and so unique. So that's kind of one of my secrets is, whenever I can, I will just have no quantise on anything, and go for more feeling."

But Sounwave's contributions didn't stop at the beats. He added "little structural kind of things", as Balouzian puts it. "It was this interesting thing where he gave us our space to feel comfortable and work alone in there. And then when it was his time, we'd go out of the room and give him, like, four or five hours to really just work. Then we'd come in and shape it. It was a really fun, free process."

Within his Mac laptop, Sounwave tends to use Native Instruments libraries. "It's just infinite sounds in there. Also,

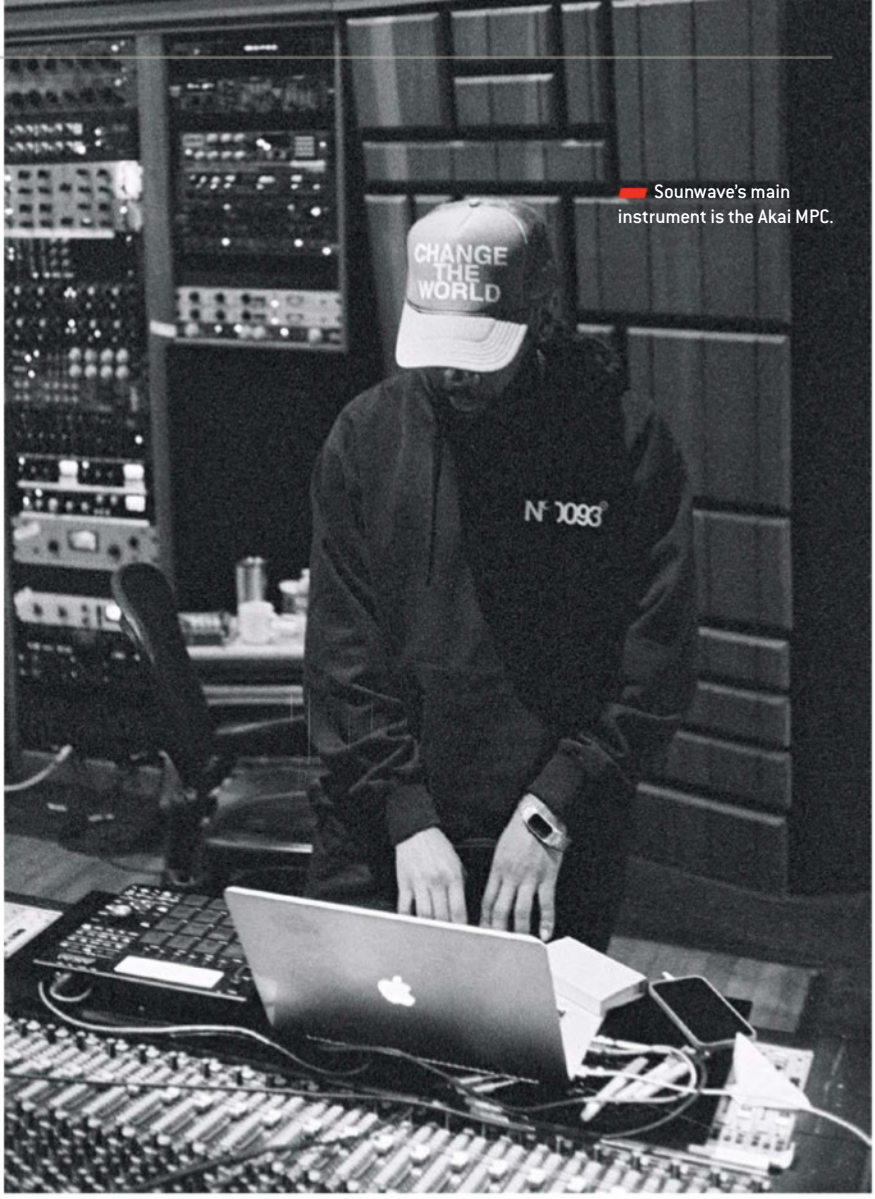
Ari Balouzian lays down a keyboard part in the live room at Conway.

Game Music

As a teenager, producer Sounwave (real name Mark Anthony Spears) was inspired by Timbaland's beats, and began making his own using a Korg drum machine borrowed from his father, before experimenting in loop-based track creation with the MTV Music Generator program for PlayStation.

"Somebody had the genius idea of creating a video game that allowed you to make music," he says. "For somebody who didn't have money — I couldn't afford the big MPC machines and stuff like that — it was great. At that point, I was like, 'Oh yeah, music is my life.'"

One demo track Sounwave made on Music Generator with rapper Bishop Lamont ended up picking up local radio plays in California. "From a video game," he laughs. "That blows my mind."



Sounwave's main instrument is the Akai MPC.

I use a lot of the [*Spectrasonics*] Omnisphere sounds. And I just mess with the EQs on that and try to get as trippy as possible. The trippier the better for me."

World On A String

At Conway, the team experimented further with Ari Balouzian's one-man string section. "We brought in a cello player at one point," he says. "But for the most part, I do it all: cello, violin, viola."

"We had three microphones set up for the strings for Ari to stack himself up," says Nasci. "So we had two [*Neumann*] U67s, left to right, and then an M49 in the middle, all on the same plane, just to keep everything in phase. Then as far as panning, it was kind of Ari panning himself. When he was playing viola, I would mute the M49 to thin it out.

"I wasn't doing too much EQ'ing. One, because I just didn't have time to go through and try to EQ all these tracks, and two, just because it already sounded pretty good. He was able to balance himself pretty well. When we were doing some cello passes, we put up a [*Neumann*] KM54, which sounds great on cello. That mic's incredible. So we used that to get a little more directionality from it.

"There wasn't too much processing on the strings when we were tracking them. I made a string bus with maybe some Fairchild plug-in, maybe some [*Softube*] Curve Bender to sweeten it up a little bit, but nothing crazy."

A similar, open- and multi-miked approach was used on some of Jacob Lusk's layered backing vocals, most notably on the gospel choir-like end section of the album's closing track, 'Mama'. "He was layering everything on just one [*Neumann*] U47," Balouzian remembers. "And we were struggling to feel that depth of how big it gets when he's singing loud. It just felt a little thin and like all these close layers. So eventually we tried putting him in different positions around stereo mics, and he just backed off and would move around like a choir would."

"I think we spent a good hour and a half, just stacking him up," says Nasci, "and we'd slowly fade in clapping, like having this church experience. It was really incredible. That was one of my favourite moments, for sure, was making that kind of emotional church choir with Jacob."

A Real Voice

Jacob Lusk's vocals were recorded using a Neumann U67 during the early sessions at Balouzian's home studio. "But then we realised if he's singing really loud, you need to have like a really controlled, clear vocal," Balouzian says. "It was kind of best to work with him for the final vocals in these more treated rooms. So, we started using a U47."

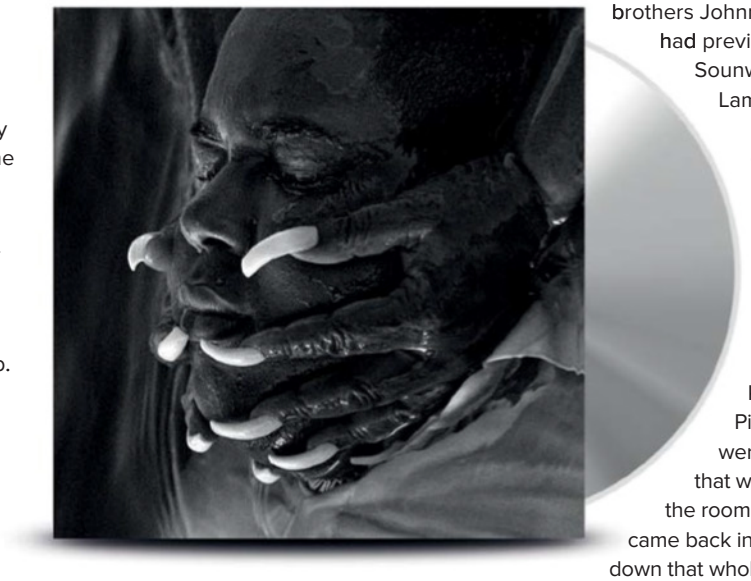
"The guys had done a lot of work," Ryan Nasci adds, "and they were saying that the U47 was the one for him. I was like, 'All right, I'm not gonna question that.' So we did a U47 into a Neve 1081 [*preamp*] into a [*Tube-Tech*] CL1B [*compressor*]. Pretty classic. On one song, the intro of 'Mama', we did use a [*RCA*] 44."

"Mostly it's real, live performances," Balouzian points out. "I love that, in 'The Great Wind', at the end, he's out of breath. At first, we were like, 'Oh, do we just comp that in?' But there's something about that where you know he made it through and that was a real take. We did that at Sunset Sound, and he sang that with the band."

On 'Love And Hate In A Different Time', Lusk's voice in the choruses was fed through a Leslie speaker, while on 'Taboo', it was routed through an Ampex ATR-102 tape machine to add slapback echo. »

» "The vocals for 'Taboo' I wanted to sound like an old Nina Simone record," says Nasci. "So we just ran them through at seven and a half IPS on that, to get them really, really gritty and midrangey and take all the fidelity out of it. I mean, it actually still ended up sounding great, because those ATRs are incredible. So, a lot of tape. A lot of slap.

"I have a little [*MicMix*] Master Room mono spring reverb that I brought. That's a great little reverb. We experimented a little bit with an [*Eventide*] H9000, which Ryan brought in. He really likes that unit. But yeah, it wasn't super-effected. I would add, just to bring it into the modern world, [*Waves*] R-Vox. But not crushing it."



Too Weird

Ryan Hope recalls there being many standout moments when recording Lusk's vocals for the album. "The last song we did was 'Taboo'. We had 'Taboo' as this instrumental that Ari and I loved. We were told, 'Maybe it's too weird... it's not a single.' We cleared everybody out the studio and Jacob just cut it. Those first takes of him singing were unbelievable."

"'Taboo' we actually wrote in Conway," says Balouzian. 'I didn't know if it was going to really work for Gabriels because it was a song in 7/8. I laid that down with Trevor Estes, who's a great drummer. He has all this great percussion and knowledge of stuff. We wrote the song, with the vocals, in Conway and Sounwave really liked it. So he laid some sounds with his drums that fit within them. Really subtle things, but they made a huge difference. It's funny, even that little woodblock sound in the chorus, I was like, 'This is why he gets paid the big bucks.' It gave it this real cinematic thing."

"'Taboo' was the one that was started from scratch," Sounwave says. "We all just locked in and just vibed out and created this. I don't even know what genre to put that song in. It's like soul orchestra/church/futuristic. That one was exciting to make. We just had so many different elements in one gumbo pot, so many minds just trying to figure out what the hell we were doing. And it turned out amazing. It's one of my favourites."

Turning Professional

At the opposite end of the spectrum from their experiments with digital beats and textures, sometimes the early stages of tracking simply involved the musicians playing together as a band unit. The '40s-sounding 'Professional' was a case in point: it segues into 'We Will Remember', which uses the song 'The Way We Were', made famous by Barbra Streisand.

"'Professional' was something we developed live," says Balouzian. "Jacob had the idea when we were in one of the rehearsals: 'I feel like singing this Barbra Streisand song over this beat.' He started singing it, and then he made that arrangement with our keys player who's also a co-producer on the song, Sam Beste. They kind of figured out an arrangement for that middle section of the Barbra part. That became kind of a highlight of our live sets.

"We did it where we recorded it like a jazz recording. We had Sam and Max Whipple, another major collaborator who plays bass and did all the horn arrangements on the album. Him, Sam and Jacob cut those two verses into the kind of beat section. They played live and Jacob sang the whole take live with them while they were playing."

Inevitably, perhaps, the team then began to add programmed beats to the live track. "When Sounwave got involved, I was like, 'It would be amazing to hear what he would do drums-wise on those parts where the beat came in...'"

Sea Sounds

When it came to mixing, Gabriels and Sounwave decided to bring outside ears into the project. Beach Noise, a trio of producers involving Matt Schaeffer and brothers Johnny and Jake Kosich, had previously worked with Sounwave on Kendrick Lamar's albums.

"Yeah, they started off as our interns for Kendrick," he says. "We had no idea *they even produced* or had a group, until we were looking for a guitar player for the end of 'King Kunta' [*from 2015's* To Pimp A Butterfly]. We were listing the guitarists that we know, and we left the room to get lunch. We came back in, and Matt had laid down that whole outro by himself. We had no idea he played guitar. And ever since then it was like, 'Now you with us on this side, too.'

"We've been rocking with them ever since. Whenever I have a project, I'm always gonna have them involved in some kind of way. Whether it's mixing, whether it's production. Like, their genius level of mind, the way they think, it's just unmatched."

"Any time Sounwave reaches out to us, no matter what, we're gonna say yes!" says Jake Kosich. "But we checked out Gabriels and we were blown away."

Mixing for *Angels & Queens* took place in Conway's Studio C, with its 68-channel Neve 88R desk, which, in fact, Beach Noise chose not to use. "We didn't really mix it on the console," says Matt Schaeffer, "but they have a lot of cool outboard gear in that room. The ATR-102 was ready to go, so we could mess with different speeds for different sounds. They have an EMT 250 [*reverb*] in there... nice, nice gear.

"We brought our [*Roland*] Space Echo, and I brought my rack of Coil mic pres, which are like '50s style. And I was using [*undecoded*] Dolby [*noise reduction, used as a treble enhancer*] on all the vocals as a parallel. So I would just send tracks out of Pro Tools through all that stuff, get a good blend of things and then print it back and mix that."

Mixing began with Lusk's vocals. "I ran all the vocals through Coil mic pres," Schaeffer says. "They have a pad so you can run line level into them to get a cool tube sound. There's

Angels & Queens was mixed by production team Beach Noise. Left to right: Johnny Kosich, Matt Schaeffer, Jake Kosich.

a negative feedback control that is kind of like an EQ, but it's more really a broad tone-shaping thing, kind of like a Pultec. So I ran all the vocals through that and then also parallel to a Dolby A."

Blind Faith

According to Jake Kosich, the trickiest track to nail was 'The Blind'. "The direction of that one from Wave and Gabriels was just, 'Go weird on this one.'" says. "We just started experimenting with stuff. Matt printed some vocals and then we put it out on a MIDI keyboard, chopped it up. Then just played a bunch of chop sounds and delays and crazy stuff. So, I think that one sticks out to me a lot because we definitely were given pretty much free rein on just going crazy on that mix."

"I sent some of the drums out to the ATR as a tape delay, but it was feeding back into the track," says Schaeffer. "So it was like a feedback loop, and I was just controlling it with the send in Pro Tools. I just played the fader like a filter on a synth. Then I also reversed the entire vocal track and did the same thing with the send on that through the ATR. So there's two tracks of feedback delay and one of them is the whole vocal track just playing backwards like an automated feedback loop.

"Then on that one we reamped a bunch of it. The drums we sent back out over the speakers in the live room at Conway and recorded the drums back in again and just slammed them with compressors."

Schaeffer, for his part, reckons that the album's opening track, 'Offering', was the toughest to mix. "There was an upright bass in there and an open-sounding kick, like it didn't have a pillow or a blanket inside of it. It was just like a concert bass drum type of sound that had a lot of low end, and so did the upright bass. And it was pretty tough getting those to sit with each other."

"'Angels & Queens' was a tough one, too," says Sounwave. "I couldn't get the drums to slap the way I wanted to. And so I had to replace the kick a few times. I tried about six or seven different kicks, and none of them fitted. It just stood out; either it was too low or too high, too weird. Then eventually I had to get a real kick and just beat that up to make it feel a little bit more synthetic. But that was the way to go."

Journeying

Looking to the future, and a second Gabriels album, Ari Balouzian and Ryan Hope both say their approach has been changed by working with Sounwave on *Angels & Queens*. "We both got MPCs thanks to him," says Balouzian. "I guess Ryan and I have brought back the drum machines a little bit. We've actually been making a lot of our own drum sounds and putting them in MPCs, and there's more percussion."

Exploratory sessions for the next album have already begun, with Gabriels having just spent a week at Rick Rubin's Shangri-La Studio in Malibu. "Usually our songs don't really go on long journeys," says Hope. "The real nuggets of them are caught and captured fast and they're kind of finished fast. We'll sit on them for a long time, and they'll get finessed sonically, but not hugely. The identity of it is usually there in the original demo."

Closing the creative circle back to where Gabriels began, Hope and Balouzian have been working on a TV score, which will likely inspire their next songs. "We're working on a score for a limited series on HBO," says the latter. "That's been kind of interesting. A lot of times for the album, songs came out of writing from earlier scores. There would be themes and melodies and chords and stuff that we pulled in. That's happening right now, as well. Stuff will get made for the score, and then we'll be like, 'Oh, shit, it doesn't really fit in the score, but it would be a really good song...'"

Cristobal Tapia de Veer

Composer & Producer

Photo: Pierre Barilier

Cristobal Tapia de Veer has become a first-call Hollywood composer — by ignoring Hollywood's rules.

SONAL D'SILVA

"I make music because of necessity," says Chilean-born Canadian composer Cristobal Tapia de Veer. "To me, it is a way of talking. I talk better with music than I do with words, so explaining music is not something that I like that much. I mean, I'm happy to talk about it, but if I had a choice I would rather not explain music."

Tapia de Veer has found himself having to explain his music a lot since 2021, when a project he scored became an unexpected worldwide smash hit. HBO's *The White Lotus* fuelled pop culture discourse and sparked a mid-career renaissance for the actress Jennifer Coolidge. Originally intended as a one-off mini-series, the show is currently filming its highly anticipated third season in Thailand, and has put Tapia de Veer firmly in the spotlight for his score, with its heady mix of otherworldly human voices, primal beats and hints of sounds that may or may not have come from a jungle. For the Sicily-set season two, his theme song 'Aloha' was given an operatic makeover, has since become a legitimate dance hit, and was even remixed by DJ Tiësto.

The composer insists, though, that there was no magic formula behind the theme's success. "Somehow I feel like when we talk about things in interviews, it feels premeditated — like I had this major plan about composing this big piece — but I have to go back and understand what I did, because I wasn't thinking at all while doing it. Ninety percent of what I do comes from improvisation: from doing stuff, making errors and then realising that something is amazing and following that path. I'm just following the rabbit! You realise what you were doing afterwards and then you could write a PhD about that stuff, but it feels like it's made up, in the sense that you really have to make a story to be understandable to the world. The stories come afterwards, and it's a result of having done something right."

Learning The Ropes

Tapia de Veer has been doing something right with music since he scored the

Cristobal Tapia de Veer coaxes new sounds from a heavily modified broken guitar.

cult British hit *Utopia* in 2013. You can hear the hallmarks of his style in the score: otherworldly vocals, unconventional instruments and dancefloor-ready arrangements that underscore the subject matter. Tapia de Veer looks back fondly on his time working on the series that served as the breakthrough for his composing career: "I had done one show before with the director Marc Munden — a period drama for the BBC called *The Crimson Petal And The White* — and the same director called me to work on *Utopia*. I think of it as a science experiment where I was in the laboratory doing something, and it exploded and that presented me to the world, I guess."

Working on *Utopia* was an educational experience. Tapia de Veer was already a classically trained musician, having graduated with a Masters degree from the prestigious Conservatoire de Musique de Québec, but composing for TV involved acquiring a different set of skills. "I didn't study to compose to the image, so I didn't know any rules, really. To me, it's like with any musician: if you give them images, they're going to start vibing, maybe on a guitar or whatever instrument is at hand, and see how that feels, how that looks with the image and that's really the only thing that I knew. There was lots of trial and error but I learned with the director, who would tell me what worked and what didn't. That was kind of my school, I suppose."

He learned not just about making music for visuals, but also about dealing with all the other stakeholders in a production. "The first time I went to London to work on the show, a line producer called me to say, 'OK, you need to meet with the sound effects guys and see that all the sounds you guys are making are going to work together.' And I told her, 'Well, I don't have the time for that, I have so much to do!'" He laughs at the memory now and at his own naïveté: "I was like: anyhow, they have to adapt to the music. It's not like I want to change the music to incorporate door sounds! In my mind this was obvious, but I learned that it's not like that. I learned on the field to deal with people, with the production, and how to make music for images."

Utopian Vision

After the success of *Utopia*, Tapia de Veer says the phone never stopped ringing. Over the next few years he scored critically acclaimed shows like *Humans*, *Dirk Gently's Holistic Detective Agency*, *Black Mirror*, and last year's big hit horror movie *Smile*, to name a few. The pattern that runs through his filmography is collaboration with directors who have an auteur-like approach to their work, which allows the score to have a strong and individual voice. "The thing that I look for the most while choosing a project is whether it feels like people understand what I can bring to a project, and whether they're going to let me help the project. I guess those are the two elements where things could go wrong. I'm not very good at making something like the temp score; I don't think anybody can do something like the temp, in the sense that it's so specific.

"People tell you, 'We don't want you to copy the temp score, but it has this feeling that we want.' That sounds great on paper, but the reality is that when you like a piece of music there is no making another piece of music like it. You really connect with that particular piece of music. Maybe it's the voice, or the singer, or the way it was recorded; there are a million things that people connect with in one piece of music, and they're going to try to get you as close to that as possible and it always ends in some kind of a rip-off, which is not a good way to work. I suppose what I'm thinking the most of is trying to be sure that I'm not going to be in those kinds of situations going into a project."

It's not easy, but Tapia de Veer has developed an understanding of the process. "I tend to work with a director for a while and then, later in the production, the director shares what we've been doing with the producers. That can be a bit of a shaky place! It can be surprising for producers when they hear the new stuff that is so different from the temp score, and I do tend to experiment a lot. Sometimes [*the producers*] are in shock about something and that can be tense, but there's a path of trying to understand where they are coming from. Most of the time they are not musicians, so quite often they don't have the language for the feedback. They get scared and if they »

» hear something they don't like, they could say, 'Can you change this little thing?' or something like that, but instead they go, 'Oh my God, this is not working!' It seems like a big deal because they don't know what to say, but it's generally not as bad as it seems and I learned that if you are patient, you're going to get to the same place that you all want to go, and everything is just stronger."

Back To The Source

Tapia de Veer's compositions come from a love of experimentation and improvisation which extends not only to his treatment of sounds, but also to the sources of the sounds themselves. "I've been recording sounds with an acoustic guitar that I found in the trash some time ago. It was probably the guitar of some kid who stopped playing at some point, so they just left it outside in the minus-40-degree weather. It was fairly destroyed so I modified it, added metal springs and things, and now there are some really interesting sounds in the instrument. You can make some really funky, spooky sounds; it's almost like a prepared piano in a way, with all the harmonics and what you can get out of the strings. You can also use it for percussion and all kinds of other weird stuff. I'm scoring a movie using this broken guitar, and the directors are also completely in love with the sounds coming from it. It's very interesting to me and I like the fact that it's something that came from the trash, because instruments can be so, so expensive and it always feels like you need to spend so much money to make good sounds."

The tradition of collecting and modifying instruments goes back to his childhood, he explains: "I didn't have instruments when I was a kid in Chile, so I experimented a lot. I had an old Spanish guitar from my parents that I began experimenting with; I started making drums out of boxes and plastic bags. I found that if you combine a lot of plastic bags to make a ball, put it on a stand and hit it with a stick, it sounds like a closed hi-hat.

"I also had an uncle who was an electrician and he told me that I could use an old speaker as a microphone. So I took a speaker, taped it on my guitar, plugged the two wires into the input microphone of a ghettoblaster that I had, and a really interesting blues sound came out of that. I could also use the speaker on a hardwood floor — if you tapped your feet on the floor, the speaker would amplify that into the ghettoblaster and, with the volume turned up, the speakers would distort and you'd have a really interesting hip-hop-style dry, distorted kick."

One of the more conventional instruments in Tapia de Veer's studio is this orthodox drum kit.

Tapia de Veer experimented with whatever was available, and he says it remains an interesting way to find new and unexpected sounds, even though he now has instruments from all around the world. "I don't know exactly how they work, but I find a way of making them work for me, in the same way as if I had made these instruments. Also, I don't have eight hours a day to really learn one particular instrument, so I have to find a way of playing them confidently. I know what a confident sound sounds like, when a musician knows what they're doing! I know how to do it with what I've studied, so I try to keep that attitude in an instrument that I don't know at all. I find certain things that I can play like I'm a pro and just use that. That's it. I don't need to be any more of a virtuoso with that instrument, it's really about the palette of sounds and I use whatever I need for the composition."

On The Phone

Like modern-day musicians everywhere, Cristobal Tapia de Veer reaches for his phone when inspiration strikes, recording musical ideas before they disappear and even jamming with himself as he records loops and builds upon an idea. "I put the phone on my leg and play into it, which is not optimal but it's amazing how fast things go if I can jam with myself on a track and keep adding to it. It feels really spontaneous, which I really like. I remember one day, I made around 45 tracks like that. It was ridiculous and I realised that I needed to find a way to make this system professional!

"Right now, I'm trying to replicate that in the studio, so I'm aiming to have all the mics ready to record, where every mic has a chain with a preamp, a compressor, an EQ and maybe even distortion and hardware reverbs and things like that. I would like to know every mic, to choose if I'm going for a brighter sound or a darker sound or a weird sound, and to be as fast as possible, so that when I record the sounds it's almost mixed and I don't have to think about plug-ins or anything like that — I am already very close to something that I like. Since I also produce and mix everything, I can take those risks of recording sound that is completely produced. I have an idea of where I'm going, so it's not a huge risk, really. That is a system that I really want to perfect: to be able to jam and record ideas very fast."

The main obstacle he comes up against is the difficulty of recreating the feel of the phone recordings in the more professional setup. "These tracks that I made on my phone didn't sound

For Cristobal Tapia de Veer, the important thing in getting usable sounds from almost any instrument is to attack it with confidence, even if you're not an experienced player. The dununs in this photo are African rope-tuned drums.

amazing, but the interesting thing is that when I imported everything into my DAW and thought, 'OK, now I'm going to re-record everything better,' I couldn't make anything better! It was just impossible, and I have the tube mics and everything! I would listen to my phone demo, be amazed and try to imitate it. At one point, I thought maybe I have to put the mic on my leg where my phone was, and it just became ridiculous. I realise there's something about when you capture an interesting idea in the moment. It sounds even technically better in a way that I could not improve with better gear, strangely enough. It really puts into perspective all these ideas we have about what kind of gear you need to make sounds that you like, but it seems like capturing the right moment is everything, really."

It's All In The Voice

Even though there are no lyrics, the human voice frequently takes centre stage as the element around which the rest of Tapia de Veer's arrangements are built. "I suppose I just realised at some point, while experimenting with voices, that there is no sound that feels more intimate than the human voice. There are instruments, like the violin or cello, that — with a really good player — can take you somewhat close to the emotional impact that the human voice has. But in the end, we like those instruments because they're close to the human voice, and ultimately the voice is a really important thing.

"That's one aspect of it, and the other is that because it is so personal and intimate and warm, when you start messing with it, it can be weirdly unsettling. The thing is, though, what I really like is to not process the voices a lot, so that they really feel natural except for something weird that's happening. You can tell there's something weird, that it feels alien, but it doesn't sound like a robot: there's no Auto-Tune, there's no vocoder, there are no voice-changing algorithms. It really sounds like someone, but like someone weird.

"In *The White Lotus*, I felt like the voices are so special that I was happy making a dance track and just chasing that energy to build throughout the track. The voices are getting bigger and bigger and I'm trying to somehow chase a sentiment of being overwhelmed »

Free Agent

Cristobal Tapia de Veer has achieved both commercial and critical success. He's an in-demand composer with Emmy, BAFTA and ASCAP awards under his belt and his works are firmly entrenched in pop culture. Yet, almost uniquely, he has chosen to work in Hollywood without an agent, a decision that's unheard-of for a composer of his calibre. His reasoning is based in practicality. "It has somehow become a feature, with time, I guess, but the thing is I met all the big agents in Hollywood. I met some really great people, some people that I would like to work with, but I've been working non-stop since I did *Utopia*. The phone never stops ringing, and that somehow put me on this path. A director asked me to be personal with the score, and because we did that at the beginning, people started calling me to do that, to be original and to be different. It's pretty much a dream come true, in the sense that I don't feel like I'm playing games too much.

"There's a system with agencies, and there's a way of thinking about doing the biggest projects and many other elements, and I suppose maybe I am a little bit wary of where that could take me. Maybe it's just paranoia, but it feels like I don't want to delude myself about what I'm doing.

"I'm not saying it's never going to happen [*that I get an agent*], but for now I like that when I go to Hollywood I'm like this guy who comes from another planet and is somehow working with everybody, but then I go away and disappear. I'm happy to feel a little bit disconnected from how things are done in Hollywood!"

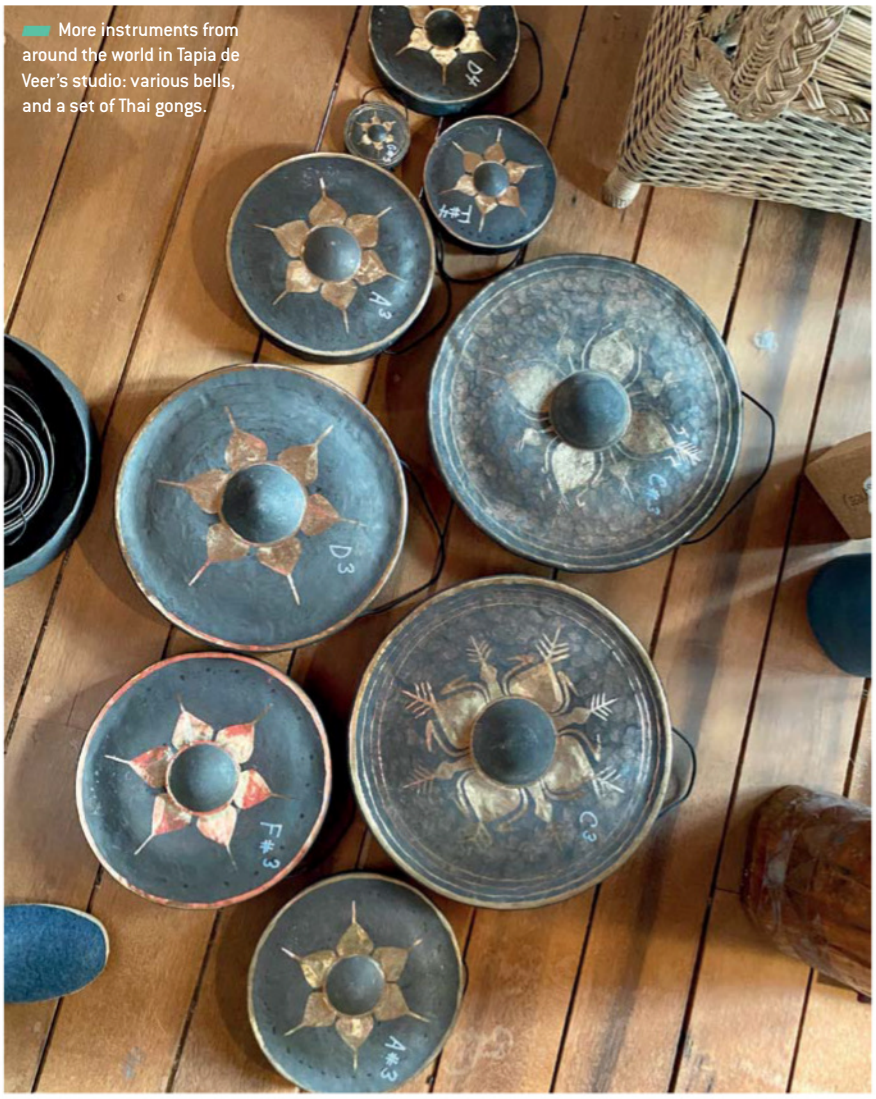
» or being at a festival or a club and completely tripping. I was chasing that euphoria in the music."

Bijou Boutique

Tapia de Veer's studio setup is all-analogue apart from Ableton Live, running on a laptop, which he says allows him the flexibility he needs while composing. "To me it's about the workflow. Ableton feels like it is really elastic, and with Pro Tools and Logic, I felt like I was trapped. To give you an example, in Live I can play with the pitch of a wave like this [*makes an up and down dragging motion*] and get somewhere, but in Pro Tools I need to open a menu, then write '2 semitones', print it and listen, and then it's possibly not the right pitch so I have to undo it... This makes no sense to me, it's going to take a year! So for me, that's the gold in Ableton because it's more than a studio. I suppose if you see Pro Tools as a studio it might be great but I'm writing with Ableton, in the sense of stretching and warping and doing all that in a second. To me, that's really important, because I need to hear all those differences and make choices. It's more interesting that way."

His studio is populated with gear that he's been collecting over the years: preamps, compressors, saturation units, cassette recorders and bigger reel-to-reel machines. He has also become a big supporter of smaller pro audio companies that make quality products and sell directly to customers at reasonable prices. "Stam Audio, for example, are making the most accurate recreations of vintage gear, so you either pay $20,000 for a mic or you could pay $1000 for a mic from Stam Audio. I just bought an 87 tube mic from Stam Audio for about $800 that I've really been enjoying. It's just crazy that we have access to this kind of quality at that price, because it feels like not so long ago you needed to spend like $4000 for that kind of quality. It's the same for gear from a company like Audioscape; there's also Locomotive Audio, from where I bought some tube preamps that look like they were made in the '50s."

His collection also includes the Dramastic Audio Obsidian Compressor, a classic SSL-style compressor, which he says sounds amazing on the mix bus and has the capacity to be pushed hard if one so desires. Tapia de Veer has been using it for years and delightedly explains: "There's



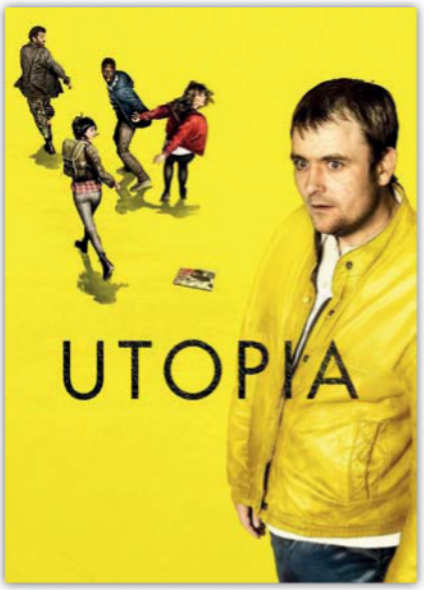
More instruments from around the world in Tapia de Veer's studio: various bells, and a set of Thai gongs.

a position on the ratio knob, indicated by a skull and two bones, and I don't know what that's doing, but it's super violent. It's a really aggressive kind of compression and distortion that I really like."

Also a regular in his effects chain is the Black Box HG-2 distortion and saturation unit. He liked it so much he tried out the plug-in as well, which he admits doesn't have the same effect — quite literally. "I have the real thing, and for me, it sounds like day and night as compared to the plug-in. I feel that we haven't come to a place where there's a clear way to test analogue versus digital. When you hear a demo of a plug-in that emulates, for example, an 1176, you might hear a snare sound with the plug-in and it feels like, 'OK, this is close enough. Why spend thousands of dollars on the hardware?' but I don't know if it actually works like that. It feels like you need to listen to an entire song and see how you feel at the end of the song, see whether your ears are getting tired with the digital aliasing and things like that, which you might not realise just by listening to a snare sound. You can do something with these plug-ins for sure, but you could also be so wrong and then you're destroying everything. With hardware, the sweet spot feels very large; it's very forgiving."

As far as the signature element in his compositions goes — the human voice — he has a specific goal and uses minimal effects to achieve it. "There is an edge where things sound really natural and otherworldly at the same time, and that to me is the sweet spot where I like to make music with the voices.

"I think my voices are always either super-dry or have lots of reverb. I have the Alesis MidiVerb; there's something really interesting about those '80s units, they glue to the sound more. Besides that, I suppose saturation is something I use a lot. I recently discovered a new company from Holland called Singular Audio, from where I found a module for the 500-series rack that is very similar to the Black Box that I mentioned earlier. It's called Tubedrve and it's this little unit that has four tubes in it. It's a really incredible unit at a great price, and with it, I feel like I can do all the saturation I need. Once again, it's that vibey sound I'm going for: not necessarily that I need to hear distortion, it's just to make things bigger and more interesting. The interesting thing to me about all these old-school



machines is that you can push them and the sounds become rounder."

The quest for a rich, warm sound means that he's less fond of cleaner-sounding modern gear, which he says he might like more if he were recording to tape all the time. "Computers feel very peaky and harsh to the ears. It's just not a natural sound, and it really feels like you need the tube gear, the distortion and all the saturation you can get to have the records sound good and warm, like they did before. When digital came out, maybe at the end of the '80s and beginning of the '90s, people seemed really brainwashed, saying things like, 'Now we can hear music, we can hear the detail.' But it's so strange that it took a while for people to realise that there was something wrong, something missing from it.

"There's so much talk about the warm sound and harmonics, like it's a choice, and I'm not sure that it's really that much of a choice — in the sense that the human ear compresses naturally. When you go to a show or there's a loud sound, your ears turn down the sound. This action of turning down sound is a familiarity we have. Pretty much anybody likes fat music, music that feels like you're close to it, and that happens with that saturation and compression. When there's none of that, it feels like daggers coming at you, like it's somehow attacking your ears. In the '90s, people would try to make things loud with limiters and digital stuff but that doesn't work that well, it's not really a nice sound. Now we've come full circle: you can buy gear that is pretty much like it was in the '50s or '60s and you can get this sound that is just more natural to the way we hear things, and I really appreciate that."

Silence Is Golden

Despite the fact that scoring is Cristobal Tapia de Veer's business, he's critical of filmed entertainment's tendency to overuse music. "In a silent film in the '20s, you did need help [*from the music*] because nobody was speaking, so you had to be super descriptive about every moment and everything that was happening. To this day, this is what film music sometimes feels like to me. It's way too detailed. When it's well done it can be really nice but the problem is, most of the time, it feels like there's someone mansplaining the movie to you.

"Silence is so important in a movie; things can be much more dramatic without music, I think, because then it feels real. I mean, in real life, when something bad happens, you are scared and it's not cool in any way, shape or form. With music, I feel like movies can lie about how difficult situations can feel in real life. Something could seem a little bit cool even if it's actually not cool at all. It can be completely confusing and it can be dangerous in that you can make things cool that should not be cool. You don't get much silence in American movies, and European movies sometimes can be the opposite. There's no music at all, and it's so hardcore because there's really nobody to help you. There should be a middle ground."

Classic TRACKS

White Town 'Your Woman'

Recorded completely at home, Jyoti Mishra's irresistibly catchy hit single 'Your Woman' inspired a generation of bedroom producers.

TOM DOYLE

Making release-ready music at home is now commonplace. But it wasn't always so. Not until the late 1980s, with the proliferation of affordable samplers and MIDI-equipped computers, often used in conjunction with cassette multitracks, were a generation of (typically rave-minded) musicians able to produce finished tracks outside of the traditional studio environment. This in turn gave birth to the concept of the bedroom hit.

In 1997, Derby-based Jyoti Mishra aka White Town took the idea to a whole other level by scoring the first bedroom-recorded UK number one with 'Your Woman' (also a chart-topper in Iceland, Israel and Spain, as well as a top five hit in Canada and top 30 hit in the States). Based around a 1930s trumpet sample motif, a loping '70s funk-turned-hip-hop beat and a bouncy synth bass line, all topped by Mishra's grainy, emotive pop vocal, 'Your Woman' was not only a distinctive blend of influences, but also insanely catchy.

"One of the things about 'Your Woman' is I designed the production to sound old when you first heard it," Mishra tells *SOS* today. "I wanted to have a track that was like a mishmash of time zones... so you weren't quite sure what year it was."

Originally championed by DJ Mark Radcliffe on Radio 1, 'Your Woman' took Mishra from indie obscurity to pop stardom (and back) with incredible swiftness. But it has since enjoyed a long life as a now-classic track, being sampled or covered by (or even simply influencing) the likes of Dua Lipa, Tyler James and Naughty Boy, as well as sounding like it might have been the sonic blueprint for Gorillaz.

Jyoti Mishra, photographed for *Sound On Sound* in 1997.

The experience of having a one-off chart-topping hit was also to teach Mishra much about what he liked and didn't like about fame, making him ultimately realise that he was happiest operating in the indie world.

"I've always felt," he says, "like I'm on the outside looking in on the scene."

One Man Band

As a kid, Jyoti Mishra's ears were always drawn to any records that featured electronic sounds, whether it be the synth bass on Sailor's 1975 hit 'A Glass Of Champagne', the eerie hooks of Jean-Michel Jarre's 'Oxygène (Part IV)', or the futuristic pop of French band Space's 'Magic Fly' (both 1977).

Mishra himself began recording at home as a teenager in the early '80s, messing around with an organ and two cassette recorders to create live-overdubbing, in-the-room recordings. "I was just amazed," he says. "I can still remember the thrill of putting a bass line to some chords I'd played. I know that sounds so stupid now and anybody under the age of 50 will just be like, 'What's he on about?' But it was a magical moment."

Having been bitten by the recording bug, by the mid-'80s he'd progressed to

Artist: White Town
Track: 'Your Woman'
Producer: Jyoti Mishra
Label: EMI
Year: 1996

using a Fostex X-15 cassette four-track and Roland MC-202 MicroComposer synthesizer/sequencer.

"The X-15 was like a quantum jump in sound quality," he recalls. "I managed to get an MC-202 off some bloke, who didn't know what it was, for about 50 quid. And that had FSK sync code. So if I gave up a track of the four-track, I could run things in sync... I could sync drum machines, and maybe not put them down 'til the final pass to a different cassette."

At the end of the '80s, Mishra became more enamoured by indie rock, after seeing a Pixies show at Nottingham's Rock City in 1988. "I just remember thinking, 'I love this,' and the way it feels when your whole body's vibrating. I was like, 'I need to do this.'

So I got this guitar from a friend... this acoustic guitar for 15 quid. The action was like the fucking Golden Gate Bridge, it was ridiculous.

"But I basically put on the Wedding Present album, *George Best*, and tried to play along with it. I just kept playing and playing, and I thought, 'I need to do a new band now.'"

The new band was White Town, who formed in 1989 and drifted apart the next year, but not before recording a self-titled three-track EP, released via 1000 7-inch singles. The record caught the attention of Illinois-based indie label Parasol, for whom Mishra as an effective solo artist made a series of releases between 1991 and 1996, along with a debut album, *Socialism, Sexism & Sexuality*, in '94.

"I tried to recreate the sounds that I'd been listening to," he explains, "which was like C86 into American noise rock and a lot of shoegaze. I was trying to do that on my own with a [*Roland*] TR-707. Then gradually, after the first album, I just thought, 'Why do I have to pretend that I've got a band to constrain me? Why do I not just record stuff that I could not play live?'"

Sequencing

Mishra's crucial gear purchase that was to shape the sound of 'Your Woman' was an E-mu Emax II. "That came out in '89," he says. "So, I had to wait for the prices to come down quite a lot. I managed to get one that was being sold as new, but it was second-hand. So I managed to haggle the price down.

"But back then when you first got your first hardware sampler, that was a revelation. You think, 'Wow, now I can manipulate sound itself in a way I could never have done before.' That must have been 1994. I recorded 'Your Woman' in '95 and I'd had the Emax for a year or so by then."

Central to his setup at the time was the MIDI-friendly Atari ST computer, running the Sequencer One software (designed by Manchester company Gajits) that had been given away free on a floppy disc on the cover of a magazine.

"It was very much an introductory sequencer," Mishra remembers. "But the best thing about it was you had a rudimentary piano roll, where you didn't see any notes or anything, you had little hollow squares. And once there was music in them of any kind, they'd go from white inside to blue, or black. So they'd just be like buckets that filled up if there was any information there. It could be a modulation wheel bend, it could be anything. So you couldn't see any notes. You just knew there was something MIDI happening in there [*laughs*].

"Basically, you had cut and paste, and you could quantise, but that was about it. So I used to use it like a big tape recorder and just start it going and play something in all the way through. Cutting and pasting wasn't easy. It was like almost maths."

By this point, multitrack-wise, Mishra had moved on from the Fostex X-15 four-track to the Tascam 688 Midistudio eight-track and also invested in a pair of Mordaunt-Short MS30 monitors. "Wow, they were brilliant," Mishra says »

» of the latter. "One of the worst decisions I ever made was stopping using them.

"Back then, I would be pootling around in that room, which was a spare bedroom. But it wasn't just my studio. That was my computer room and my doing work at uni room. I'd be sitting there for hours listen to Radio 1 over the same speakers, so I'd got completely used to them. I knew exactly how they sounded for normal music. So when I was doing my music through them, they were great."

When Mishra first released 'Your Woman', it was as the lead track on a 1996 EP titled *>Abort, Retry, Fail?_*: a reference to the fact that he was at the time trying to update his system to a PC running Cubase and had suffered a hard-drive crash.

"I was just getting into PC sequencing," he groans. "I think I had a dodgy version of Cubase. Obviously once you move from an ST to PC, the sequencers get way more usable, and you have all this power... but then it crashes. And your ST never goes wrong. My fucking hard drive crashed and all these songs I have never recovered since were on this hard drive. The songs that were still on the ST, like 'Your Woman', I still had them because they were on floppy disks."

Sampling

The spark of inspiration that led to the creation of 'Your Woman' was its trumpet sample. It was lifted from the 1932 track, 'My Woman', by Lew Stone & the Monseigneur Band (featuring Al Bowlly), which Jyoti Mishra discovered on the soundtrack album of 1978 BBC musical drama series, *Pennies From Heaven*.

"I thought, 'Oh, I liked all these '30s songs... let's have a listen to it.' And then this one track came on. As soon as the trumpet came in, I was like, 'Wow, that's good. That's really catchy sounding... really spooky sounding."

Later, when 'Your Woman' became a hit, some listeners mistook the riff for John Williams' 'The Imperial March' from the *Star Wars* film series. "It is very similar," Mishra acknowledges. "But I mean, then again, you could say, 'Well, it's just arpeggiating around a minor chord. How many different ways of doing that are there?'

"Years ago, a journalist was doing some review and they said people liked the riff because it's like children at school, going 'Na na na na na'. It's hardwired into us."

The beats on the track, meanwhile, derived from a sample CD produced by Parliament/Funkadelic's George Clinton in 1993 called *Sample Some Of Disc — Sample Some Of DAT*. "I naïvely thought it meant it was royalty free," says Mishra. "No, it wasn't... so I got sued. The trumpet sound was cleared quite quickly." Key to the groove of the track was the fact that Mishra doubled the beat. "One is in strict 4/4," he explains, "and one is in swung 4/4. So they don't line up. They lollop along."

Jyoti Mishra's studio, post signing to EMI... Front and centre are the E-mu Emax II sampler and Tascam 688 cassette recorder used on 'Your Woman'. Also at least partially visible here are a Moog Rogue, Roland JX-3P and PG-200, a Novation Bass Station, Casio CZ-101 and a newly purchased Akai S-3000. Bonus points if you can spot a copy of *SOS*.

In terms of the bass line, Mishra at the time had a few synths at his disposal, including a Roland JX-3P and a Casio CZ-101, but chose — at least initially — to use his Moog Rogue. "That was my first ever monosynth," he says. "I used to gig with that live and it was an amazing sound.

"Then on 'Your Woman', I was playing around with the Moog, and it was like, 'Yeah, this is good', but it wasn't quite right. So I sampled the note. When it goes down to low C, it's really low. It sounds kind of weird down an octave... What you're hearing is the growly note of the lower sample. The actual sample itself is just a standard Moog-y bass sound, sampled around, say, Middle C and then played too low."

Along with the bass, practically everything else on the track came

from the Emax II. “The piano was just a standard Emax library sound,” Mishra says. “That would’ve just been the piano it came with. The only things that aren’t Emax II are my voice and the first time you hear that guitar, when I’m playing it. Every other time, I sampled it on the Emax. I couldn’t play it again because I was shit [*laughs*].”

The only other beat on ‘Your Woman’ is the one that briefly pops up at 3:09, which was produced by Casio’s classic (and affordable) mini-synth, the VL-1, as used by the Human League and the Fall, but made famous by German band Trio’s 1982 novelty hit, ‘Da Da Da’. “I must have got it in 1981,” Mishra remembers. “So there’s the ’81 middle eight bit with a beepy-boopy VL-1, which people would only know from Trio if they’d heard it ever.”

Recording

Similarly cheap was the microphone that Jyoti Mishra used to record his vocal for ‘Your Woman’: the Realistic PZM that was sold for £19.95 by tech chain store Tandy. Although normally powered by one AA 1.5V battery, it could be modified by using two mini 6V batteries together to significantly improve its dynamic range.

“You’re putting a stupid amount of voltage in them compared to normal,” Mishra says, “and it makes the frequency response better. Also, they’re so forgiving because the pickup pattern is ridiculous. You can clamp it into a mic stand, you can stick it on the wall, you can hang them off something and play a guitar in front of them and they will just get everything. You can do this wiring to adapt it to phantom power and you get even better signal out of it. So, I cut one of mine up and I fucked it and so I only had one [*laughs*].”

On ‘Your Woman’, Mishra recorded his vocal using the PZM routed to the Tascam 688, via a Boss BE-5 guitar effects unit (featuring compression, chorus, digital delay and overdrive/distortion). “The best thing about the BE-5 was it had a noise gate on it,” he says, “so I could record stuff through it. I mean, it’s not great audio quality, but I could record stuff through it onto my 688 and gate it at the same time. This to me was amazing.

“So, I used the BE-5 for my vocal chain, although it was a guitar pedal. And then when I was working out the vocal for ‘Your Woman,’ I was just experimenting, and I put the overdrive/distortion on it. Once it’d gone through the overdrive, it took out all the bottom, and it just had this nice telephone-y sound. It fitted because I just thought, ‘Well, it sounds like a ’30s track now.’

“But also one of my favourite ever pop singles is the Buggles’ ‘Video Killed The Radio Star’, and that’s the sound Trevor Horn has on his vocal. I just thought, ‘Oh, it sounds like the Buggles. That’s cool.’”

Mixing

Jyoti Mishra still has the cassette multitrack for ‘Your Woman’, which he’s stashed for safety in a biscuit tin. Surprisingly, perhaps, he only used five of its eight available tracks.

“One track is the FSK sync,” he says. “One track is the vocal. There’s a stereo track of all the keyboards and basically all the backing. There’s one track that’s the bass line and one track for the guitar... so five tracks.

“Then I put this other crazy synth solo on it. I did this Moog-y thing that was through the verses as a countermelody, from the second verse onwards. And I was listening back to it, and I just thought, ‘That’s shit.’ So I left off. So it would have been six tracks, but it’s five tracks.”

As a result, of course, ‘Your Woman’ didn’t take long to mix. “The thing about [*recording using*] four-tracks and eight-tracks is you’ve already committed,” says Mishra, highlighting the one benefit of having to bounce. “On ‘Your Woman’, how loud is the piano compared to the organ? It doesn’t matter because you can’t change it.

“In the starting verse, the keyboards and the organ are both centre-panned. From the second verse, I panned them left and right. I did that as it was being recorded down from the ST onto the 688. So, I can’t change that. That’s just done.”

At the time, however, the norm was to leave the Atari ST MIDI parts triggering synths and samplers running live until the mix was laid down to DAT. Mishra says that his decision to bounce the parts to tape was as a result of his calamitous hard-drive crash.

“By that stage, I had the fear of losing stuff that wasn’t committed to tape,” he laughs. “Even if it’s gone down onto a master cassette or a master DAT, it’s like, in the ether. So I was like, ‘I can put another 15 tracks of different things down as I go down to DAT. But what if I lose the sequence? What if I lose those sounds?’ I was really — and rightly so »

Somewhat overshadowed by a more recently acquired Korg Trinity, Yamaha 02R mixer and Tascam DA98 digital recorder, you can just glimpse a Roland TR-707 and a very well-hidden MC-202.

» because of the hard-drive crash — getting more paranoid.

"So I'd rather record them down as a stereo pair, like I did on 'Your Woman', and have too many things crammed into one stereo pair. At least it's there. And it has the effects on... it has any delays on and you can't change it afterwards. And that's it. If you don't like it, you could always re-record it."

Another key factor in bedroom musicians being able to finalise mixes for release was the increasing availability of DAT. Access to digital mixdown at home — in Mishra's case, using a Sony DTC-750 — meant no degradation in sound quality at the final stage.

"At first I couldn't afford a DAT machine," he remembers. "So I used to rent one from the local music shop. They'd let me rent it out for 50 quid a day or something like that. I used to do all my recording, then have a day of mastering and try and record all my things to DAT.

"Then the price came down, so I could afford a DAT. I had one by '94 definitely. When the consumer DAT machines appeared, I was like, 'This changes everything.' Provided you gated stuff on the 688 as much as you could, I could get reasonably quiet recordings. That was a huge difference."

Number One

In the summer of 1996, 'Your Woman' was featured as the opening track of the *>Abort, Retry, Fail?_* EP, released by Parasol Records. Mishra was at the time earning extra cash by DJ'ing and, one night, decided to give the track a spin.

"I put it on, and people danced to it," he recalls. "One of my friends there was completely wrecked, and he came over to me at the end of the set, and he was just going, 'Mate, I love that song with the trumpet.' I was like, 'That was my song'. He looked at me and he was like, 'Fuck off... I've heard your stuff, it's shit'. And then he just walked off.

"So when I went back home, I told my then-girlfriend about what had happened. She said, 'Well, I told you it's a really good track. You should send it off.'"

Given the pile of rejection letters that Mishra had accumulated in recent years, he was understandably reluctant to post the EP off to major labels. "But then we scraped up enough money to send off a few," he remembers. "And one of them was to Mark Radcliffe at Radio 1. If he hadn't spotted it, none of this would have happened. Nobody would know the song."

Radcliffe first began playing 'Your Woman' on his night-time Radio 1 show in the autumn of 1996, and then continued to do so when he sat in for Chris Evans on the far more high-profile breakfast show. "Daytime listeners go mad," Mishra recalls. "And then they're pestering [*other DJs*] for it. I'd phoned Parasol and said, 'You might need to press maybe 2000 more'. And then once it moved to daytime, it went absolutely mental. Woolworths were trying to ask for 40,000 copies. And I'm like, 'Well, there's 800...'"

Luckily, radio plugger Neil Ferris (whose other clients included U2 and the Spice Girls) contacted Mishra, offering to take the track to the major labels. A deal was duly struck with EMI to re-release 'Your Woman' in January 1997. "I would have been home with my girlfriend when we got a call from the EMI people saying, 'It's number one this Sunday'. It sold 400,000 copies in just over a week. I've got the gold disc upstairs in the bedroom [*laughs*]. It's not even on the wall because we've been doing decorating."

But when Mishra was invited to appear on BBC 1's flagship music show

Top Of The Pops to perform the track, he turned down the offer, unsure of exactly how — as a bedroom artist — to present the track onstage.

"There were great electronic bands already, obviously, like the Prodigy where it was like, 'Well, that's how an electronic band should be. Me, I'm just a fat bloke. I'll be there prodding a synth. That's gonna look awful.' Also, by that time, I'd experienced a little part of some celebrity. And I didn't like it."

Subsequently

Record company interference forced Mishra to release the trip-hoppy 'Wanted' (featuring singer Ann Pearson) as the follow-up White Town single, when he preferred another track, 'Undressed', the slow-burning electronic ballad opener of his 1997 album, *Women In Technology*.

"I really liked 'Wanted'," he says. "But it's very much an album track, and I said that when they chose it. They made a video and took it to Radio 1, who refused point blank to play it. They said, 'It's an album track. We're not playing that as a single.'

"So then they had to about turn and go with my original choice, which was 'Undressed', and make a video for that. And by this time, it was three months down the line. And it went into the charts at number 57... with a bullet [*laughs*].

"The album didn't do well at all. Because at the time, people really didn't like the album. Now people are saying it's a great work and I'm like, 'Thank you... you could have said this 25 years ago.'"

But as Jyoti Mishra returned to the indie world, releasing another five subsequent White Town albums — the latest being 2021's *Fairchild Semiconductor*, a soundtrack for an imaginary TV series — 'Your Woman' continued to have a life of its own.

In 2005, Amy Winehouse's friend Tyler James released a version of the song as a single from his album, *The Unlikely Lad*, and in 2011, Australian singer/songwriter Darren Hayes chose to cover 'Your Woman' live on Radio 2. A year earlier, Naughty Boy (with Wiley and Emeli Sandé) had both covered and sampled the track on 'Never Be Your Woman', while in 2021 Dua Lipa (and producer Koz) actually sampled the original Lew Stone 'My Woman' track for 'Love Again', bypassing the White Town track entirely.

Mishra insists that the latter didn't irk him especially. "It was amazing... I loved it," he enthuses. "It was a huge boost in terms of getting people to listen to 'Your Woman'. And also, because it was 25 years ago, most kids have never even heard the song. But they're gonna hear a sample, and then they go online and discover 'Your Woman'.

"But it's weird now," he adds, "because the sample itself has spread such that I don't think people connect it back to my track at all. It's just out there as itself now."

In 2017, to celebrate its 20th anniversary, Mishra reimagined and re-recorded the track as 'Your Woman 1917'. "I basically tried to do it with things that would sound age appropriate for 1917, apart from the vocals... because the vocals would never sound that good, 'cause they'd be cut to lacquer.

"Somebody said recently on YouTube, 'OK, so you reimagined this 1997 song as a 1917 song, yet you made the vocal sound like nowadays. But on the original, it sounds like the 1930s.' I was like, 'You're the only person who's worked out that I did it deliberately!' [*laughs*]."

Down the years, Jyoti Mishra has inevitably been tagged as a 'one-hit wonder'. But as he always cheerfully points out, that's way better than being a 'no-hit wonder'. "I feel privileged and lucky," he says today. "Ninety-nine percent of musicians struggle their entire lives for no reward whatsoever. Like, how many people do you know that have put 200 percent into their music, and they've tried their fucking best and they never made it? How many people do you know now that are talented motherfuckers that 30 people listen to on Spotify?

"I am a terrible musician [*laughs*]. I'm a rubbish vocalist. I'm an absolutely awful guitarist. I'm quite good at keyboards. And I would say a quite good producer. There are plenty of people better than me in loads of different ways, hugely more talented, who will never make it, never make any money. And more importantly, never connect. I feel so lucky to have had anything at all because most musicians have nothing.

"So," Mishra concludes, "to have one thing you do connect with so many people is a gift."

I AM LOVING THESE LAUTEN AUDIO LA-120 MICS.
GREG WELLS
ADELE / KATY PERRY / JOHN LEGEND

Q&A

QUESTIONS ANSWERED, PROBLEMS SOLVED

Q Should subwoofers be placed behind a desk or console?

After running the research gauntlet and consulting with *SOS* and reading Hugh's excellent review [SOS *July 2014:* www.soundonsound.com/reviews/genelec-ones], I've decided to upgrade my studio significantly with a pair of Genelec 8341 The Ones monitors. I'm probably going to run them for a while with no subwoofer, but I'm already researching what it would mean to add a sub (Genelec 7360A) and to implement it spatially in my room. In an image in Genelec's GLM manual, they are suggesting a placement that would have it (if one draws a straight line from the woofer port to the listener's ear at the listening position) sending the direct sound waves right through the underside of the mixing desk and, for me, through my substantial analogue console. Is it perhaps not an obstruction issue because the sub-80Hz frequencies aren't so directional?
***SOS* Forum Post**

Hugh Robjohns, Technical Editor

We've written about choosing and placing subwoofers a few times, and my *SOS* April 2007 article (www.soundonsound.com/sound-advice/all-you-wanted-know-about-subwoofers) is a good primer if you (or anyone else reading this) requires one.

As for the specifics of your case, the wavelengths are so long at low frequencies that the sound waves bend around most furniture as if they weren't there, so you don't need to worry about obstructions in the direct path at all. You might need to worry about the space below a desk being a resonating chamber, though — particularly if the desk has solid sides. Thankfully, one of the advantages of a sub is that you have quite a lot of latitude in where you put it (unlike the main speakers, which have to be in an equilateral triangle from the listening position). So that allows you to locate it in the optimum position relative to room modes: if you find a good spot along the wall behind the main speakers, that's great, but if it works better in the room on a side wall that's absolutely fine too. If a subwoofer is set up correctly, and it's a good, low-distortion design, then you shouldn't be able to tell where it is just by listening with your eyes shut!

Genelec's 7360A: it matters where you put a subwoofer, but in relation to the room modes rather than relative to the other speakers and the listening position.

Q Are SM57s and SM58s sensitive to phantom power?

I have a question about Shure SM57 and SM58 mics and phantom power. My Mackie 802-VLZ3 mixer's phantom switch enables phantom power for mic inputs 1-3 — it can't be switched individually for each input. If I used my Rode NT1 [*a capacitor microphone that requires phantom power to operate*] on channel 1 and, say, an SM57 [*a moving-coil dynamic mic that doesn't require phantom power*] on channel 2, would the SM57 be affected by the phantom power it would have available to it? I have read the manual, and apparently a ribbon mic would be upset!
***SOS* Forum Post**

On some smaller mixers, such as Mackie's 802-VLZ3, phantom power is switchable only globally or in banks. Thankfully, moving-coil dynamics and even most ribbon mics won't even 'realise' phantom power is present!

Hugh Robjohns

It's not an issue. The two ends of the moving-coil element in the SM57/58 are wired to XLR pins 2 and 3, and the body is wired to pin 1 (screen). Phantom power puts 48V on both pins 2 and 3, returning to pin 1. So as far as the moving-coil mic is concerned, it's got no voltage difference across the coil (because there's 48V at both ends) and as there's no connection between the coil and pin 1, no phantom current can flow. Essentially, the SM57/58 has no idea phantom is present at all. It would be a pretty poor ribbon mic if it were so easily 'upset' but there are some (mostly very ancient) ribbons that genuinely don't like phantom power. It's not something I've ever worried about, though, and I happily use Coles and AEA ribbons with phantom on the lines. In fact, the tie-lines between studio floor and control rooms at the old BBC Television Centre had 48V permanently enabled on the floor boxes, to allow mic patching on a standard (PO316 B-gauge) patchbay without having phantom across the patchbay's sockets. It also meant that all mic plugging/unplugging on the studio floor was 'hot plugging' with phantom permanently on the lines.

SOUND ON SOUND

PODCASTS

Gain Staging - Episode 2

The Recording Stage

In the second episode David Mellor focuses on gain staging during recording, by balancing mic input, preamps and gain within your DAW to achieve the optimal sound level.

Clark - Sus Dog

Recording his new album

Chris Clark, known professionally as just Clark, reveals how he got signed to Warp Records, his recording and production workflow, and how Thom Yorke ended up as a guest vocalist on his latest album, Sus Dog.

LoneLady

Punch-and-crunch electro

Created on sequencers, synths and drum machines in a basement bunker beneath Somerset House, London, LoneLady talks to Caro C about the making of her third studio album Former Things.

EastWest Composer Cloud+

Behind The Brand

Doug Rogers is President of EastWest, the Soundsonline website and owner of EastWest Studios. Here he chats to Sam Inglis about what it takes to develop their award-winning sample library releases and how you can access everything via their subscription service, Composer Cloud+.

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SAMPLE LIBRARIES

Heavyocity
Mosaic Neon
Kontakt Instrument

★★★★★

The latest addition to Heavyocity's Mosaic series is Mosaic Neon. Like its predecessors, Neon delivers a specific palette of sample-based sounds alongside the impressively powerful Mosaic front end/engine, with its three-part sound blending and plenty of options for sequencing notes, creating arpeggios and an excellent sound-modulation system. Neon's sound palette is based around vintage and analogue synth sounds and, if that's already got you thinking *Stranger Things* or *Blade Runner 2049* (or any of the multitude of other small/large-screen hits that have also trended on the vintage synth soundtrack style), then you would be very much in the right ballpark.

The core library runs to around 3.7GB of samples that, in turn, provide 80+ individual sounds that can be mixed and matched within the three-slot engine. When browsed within the UI, the sounds themselves are organised into five categories: Pads, Keys/Mallets, Chiptunes, Digital Noises and Attack Partials. As might be expected, the Pads include longer sustained sounds, while those within the Keys/Mallets category are generally shorter and more obviously useful as melodic or bass sounds. The Chiptunes sounds include some classic old-school computer sounds but, in the main, are somewhat more grown up in nature (that is, not just bleeps and bloops). The Digital Noises are... well, noisy, while the Attack Partials are mostly short impact sounds; more on these in a minute.

To get you started, Heavyocity's sound-design team have create over 100 presets using the impressive feature set of the sound engine to fully demonstrate just what the core sounds are capable of. These Kontakt presets are themselves organised into categories: Arps, Pads, Playables and Rhythmic. Within the context of the underlying sounds, these actually cover a lot of ground. As with other Mosaic titles, given the depth offered by the engine, even a single preset can go a long way in terms of writing a single musical cue. In that context, the Arps are an instant hit; just hold down a three- or four-note chord and slowly sweep the mod wheel for some hands-on sound control, and it's amazing just how quickly an idea can appear. The Pads offer plenty of evolving textures (great for creating tension), while the Playables sounds easily span mystery to magical and all stops between. The Rhythmic category is also impressive, and blending an Attack Partials with a short, down-tuned synth sound makes a great candidate for creating arpeggiated pulse-like rhythms.

If Mosaic Neon sounds like it might be right up your *Stranger Things* street, do check out Mosaic Leads as well. It's within this same sonic territory and, once you have found your way around the sound engine, there are plenty of scoring possibilities to be explored in either title. Neon is a great addition to the Mosaic range, the sounds are fabulous and, as ever, the three-part engine is both powerful and creative. A great choice for busy media composers wanting the sound of vintage and analogue synths but without the hassle of owning vintage and analogue synths! *John Walden*

$99

www.heavyocity.com

Quiet Music
Solaris GTR
Plug-in Instrument

★★★★★

Available for Windows and Mac, Solaris GTR is an inexpensive yet impressive-sounding virtual instrument that uses sampled guitar sounds as sources before processing them to create a range of delicate, piano-like textures that are a perfect fit for chillout or relaxation music. The 192kHz, 32-bit samples were created using an Ibanez AS73G electric guitar and treated, so we are informed, using pedals and plug-ins including the Empress Zoia, Hotone Ampero 2, Seymour Duncan Andromeda, Guitar Rig 6, and so on. After editing, the sounds were converted to 44.1kHz, 24-bit WAV files and occupy roughly 2.3GB of space once installed.

Once installed you'll find 35 presets where further adjustments can be made using the intuitive control set, which is divided into logical sections with just a handful of controls in each. For general shaping there's an ADSR envelope shaper augmented by a selectable low-pass or high-pass filter with variable cutoff frequency. There's an LFO modulation section with selectable source, destination and wave shape, which can be controlled by velocity, aftertouch or a mod wheel. The most obvious setup is to use the keyboard's mod wheel to bring in vibrato. There are global volume, pan and tune controls as well as separate effects modules for reverb, chorus, delay, and distortion, each with their own bypass button. The distortion section includes a wet/dry mix control and a switchable cab emulation filter. Delay can be tempo-sync'ed with different left and right channel settings.

Whenever I've sampled clean guitars, they always end up sounding more like pianos when played via a keyboard, probably because the sound is too consistent. Quiet Music have turned this apparent limitation into a positive by creating a surprisingly wide range of sounds that somehow straddle the world of pianos, often hinting at electric pianos or music boxes, sometimes morphing into delicate pad-like sustains. Some even seem to have a vocal quality. The processing that has been used gives most of the sounds a wonderfully relaxing feel, and if you're into producing music for the lentils and crystals community, Solaris should be considered a must-have. However the sounds also sit well in contemporary pop production, chillwave or ambient productions. Quiet Music have the knack of building simple instruments that create evocative, soothing sounds and personally I don't think they get the attention they deserve, especially given the ludicrously low cost of their plug-ins. And Solaris GTR? Needless to say, I really like it. *Paul White*

$30

www.quietmusic.eu

Emergence Audio

Double Bass Textures

Kontakt Instrument

★★★★½

Adopting the familiar format of a Kontakt instrument, Double Bass Textures incorporates a healthy 5.18GB of sampled content; a considerable hike on Emergence's Violin and Cello Textures, which occupy similar stylistic bandwidth. However, it's the double bass itself that is particularly exciting in this setting. It's capable of some pretty remarkable sounds, thanks to its enormous resonating cavity and potential to bow in numerous different styles. It doesn't take long to uncover some of these sound-source gems, but before I head into upright-bass-land, let's consider the Kontakt instrument itself.

The interface is simple, elegant and engaging, but very capable. The left and right of the instrument are a mirror image, allowing engagement of up to two samples; one per side. The samples can then be adapted immediately, beginning with the simplistic concept of an ADSR envelope, and extending to a number of LFO-based modulations, which are hardwired to several visible pots, namely filter cutoff, resonance and panning. You don't have to use modulation in favour of fixing values across the board. Even in this stationary stance, the potential stakes are high, thanks to the large and centrally located balance pot. Engineering a sound which creeps and evolves across the stereo field, employing just envelopes and panning, is as simple to do as it is highly effective.

Much of the quality begins at source; 16 samples populate the sample-selection drop-down, with each playing out over a considerable length of time. There are no round-robins, which isn't an entire surprise, as this is not designed to be a conventional instrument clone. It's perfect for creating drones and textures, using bowing techniques that generate harmonic content that's ripe for filtering and shaping.

The Normale sample is about as regular as the samples get, with the purity of no-vibrato and con sordino available. In both instances, there is an abstract sense of time, as the bow switches direction at irregular points. It's difficult to disguise this change of direction, as upper harmonics enter the equation with each bow reverse. Other bowed samples provide similar sustained interest, which include sul tasto, sul pont and flautando. There are similarities in some of the sonics, but as they are each played individually, you'll always find a degree of player-based irregularity. By contrast, the Punta Darco and Detached samples both play out, with marcato strokes, through a series of random note lengths and dynamics.

The filter subtracts any harmonic-based elements that you wish, in either high- or low-pass modes. It's then over to the LFO modulations to excite the content, which can be achieved at a free rate, or sync'ed across the plug-in and/or your DAW. You can engineer an awful lot of rhythmic or unravelling movement in this way, and while Emergence supply plenty of presets to get you going, it's so easy to get creative. The Init patch quickly becomes a useful diving-off point.

With great sounds and a powerfully simple interface, this is a sonically rich and worthy addition to the Textures line-up. *Dave Gale*

$99

www.emergenceaudio.com

Zero-G

Ethera Gold Atlantis 2

Kontakt Instrument

★★★★★

Those that have used one of the Ethera sample libraries may already appreciate that the sounds themselves are pretty epic, but it's now also true that the whole Ethera library series — produced by Stefano Maccarelli and Zero-G — has, itself, become fairly epic in proportions. While the focus of the whole series is on cinematic-style vocals, there is a useful PDF document available from Zero-G's website that explains the key differences between the various titles. The latest addition to the series is Ethera Gold Atlantis 2. As with the original Atlantis release, the style of the vocal samples for Atlantis 2 suits epic fantasy with a strong medieval/Celtic vibe. However, the 14GB of sample content is all new, so fans of v1 will find plenty of additional material here for the (undeniably inexpensive) upgrade price.

Indeed, there is a staggering amount of content even for those paying full price. Three different cinematic singers (Ethera regular Clara Sorace is joined by Elena Borroni and Giulia Colantonio) have supplied the raw materials and Stefano has transformed these performances into 26 different vocal instruments for the Kontakt front end. These fall into a number of different forms providing textural/pad sounds, various 'phrase builder' options (allowing you to build custom chains of syllables) and collections of vocal phrases (six different instruments — with contrasting vocal styles — and over 2000 phrases in total) that can easily be sequenced to provide extended performances. There is also a fabulous Kontakt multi featuring a combination of the individual instruments that is simply epic to play if you want to fill your sound stage with beautiful ensemble vocals.

However, for me, the undoubted highlights of Atlantis 2 are the various True Legato instruments. These combine the fabulous sampling with very clever Kontakt scripting to deliver some incredible expressive — and very playable — vocals. Each of these offers a distinctive style. The Atlantis True Legato provides the backbone of Celtic-style but you also get Romantic, Elven and Valhalla True Legato options, each of which has their own character. For example, the Valhalla instrument has a stronger, more dramatic delivery style. With multiple articulations, control over the speed of the legato, no looping, and a wonderfully natural decay, the sounds available here undoubtedly deserve to be described as 'epic'. And, with the effects options built into the instrument including an excellent reverb, you really do only need to trigger a note or two and the results can be truly magical.

Clearly, like other titles in the series, Ethera Gold Atlantis 2 is a niche product. It's firmly aimed at film and media composers so it will not be for everyone. However, if your score requires the sound of Celtic/medieval vocals, Atlantis 2 will not disappoint. Beautiful, haunting female vocals suitable for the most blockbuster of musical contexts and for a very modest price. *John Walden*

$89.95

www.zero-g.co.uk

Audio examples of this month's libraries are available at www.soundonsound.com.

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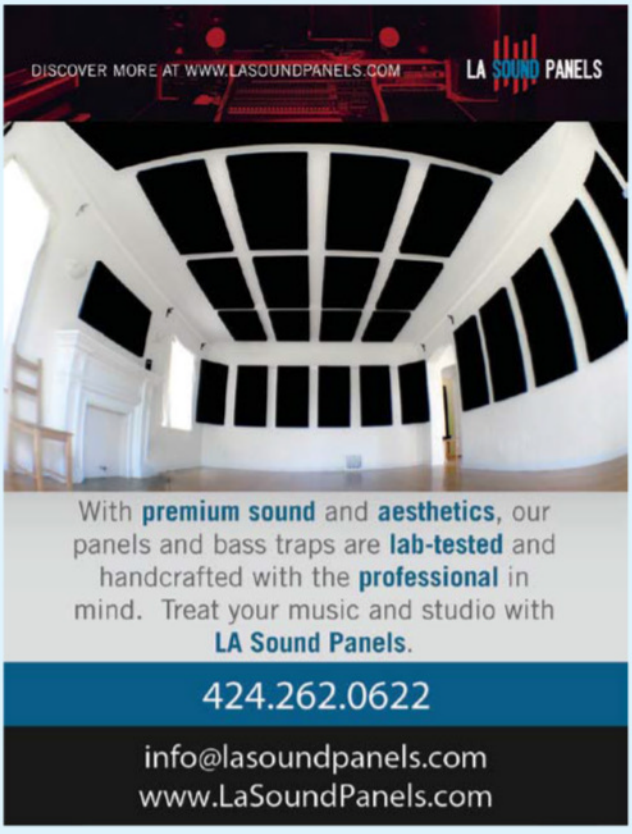

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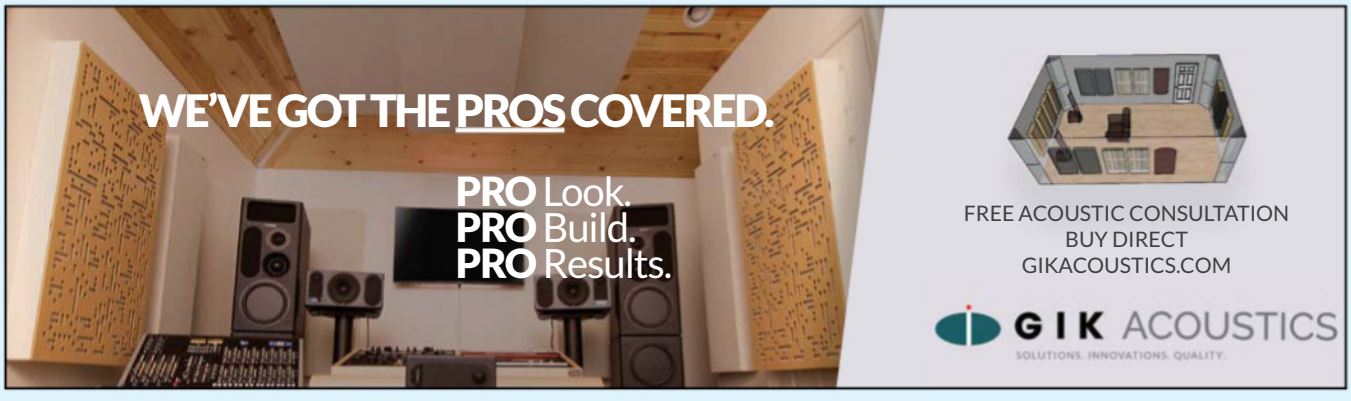

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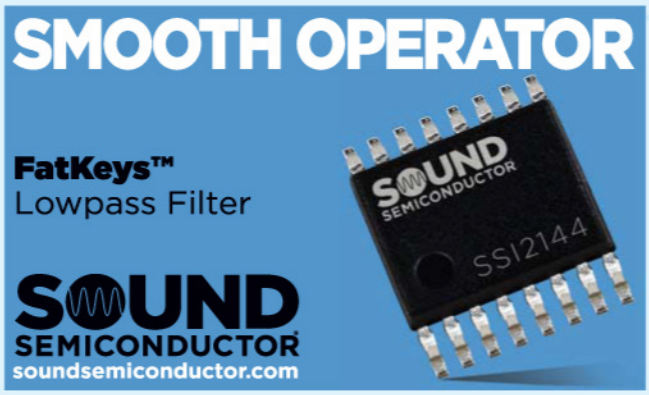
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SOS WHY I LOVE... SOUND ON SOUND

JYOTI MISHRA

I recently did an interview for *SOS*'s Classic Tracks feature, which delved into how I managed to have an international number one song from a bedroom in Derby. That song is often called one of the first charting 'bedroom pop' songs, which is debatable. What I think is more ironclad is that it's a chart hit shaped by *Sound On Sound*.

It's the early '80s. I'm a synth-obsessed teenager, worshipping Kraftwerk, Depeche Mode, Heaven 17, the Human League, Ultravox and Japan. These are my gods. And I want to make music like them. But how do you record? What is a synthesizer? Do I have to go to a studio? Nowadays, questions like these are trivial, solutions a click away. Before the web, not so much.

So I devoured magazines. *One Two Testing*, *Electronic Soundmaker*, *Electronics & Music Maker* and *Home Studio Recording*. Then, in November 1985, a new pup arrived: a very slick, colourful magazine called *Sound On Sound*. Immediately, I loved the level of writing in *SOS*. it seemed to hold itself to higher standards than other mags. It became my Bible of not only what to buy but also why to buy it and how to use it. I bought a Tandy PZM mic because of an *SOS* feature and I recorded everything with that mic for years, including the vocal on 'Your Woman'.

I didn't have much money for gear, but what I had, I spent wisely thanks to *SOS*.

Other magazines came and went and were important to me (shout out to Sequencer One free on a floppy with *ST Format*!) but *SOS* was my main source of technical info. I don't think I'm exaggerating when I say that *SOS* was my equivalent of taking a never-ending music tech degree.

The reason I'm unloading this Gen X nostalgia onto you is that these days we have myriad sources for information. Frankly, we're swamped. But using terminology I first learnt from *SOS*, the signal-to-noise ratio is low. When every other YouTuber is blatantly shilling in return for free gear, how is a kid meant to know what's good and what's merely lucrative for the reviewer? Now, more than ever, we need unbiased reviews.

I believe *SOS* still stands out. If you look at its coverage of events like Superbooth, I think you can see that difference. Obviously, there are other great sites/reviewers but when I have gear lust, the first thing I search for is the *SOS* review. It's a kind of ritual.

Now that I write for *SOS* myself, I always have teenage me in the back of my mind: small money but huge dreams. I write for that person. I'm paying it forwards. I'm excited and intrigued to hear what that person, what that next gen of self-taught bedroom producers, is going to throw at us wizened old people.

AD INDEX

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